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Electronic music instrument practice and the mechanisms
of influence between technical design, performance practice
and composition

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Abstract

This thesis examines the practices and techniques involved with particular electronic instruments and proposes an archaeological approach to reconsider the ways in which noise can communicate various details of instrument design and practice to the listener. I present two case studies concerning electronic music practice using repurposed devices - stepped filters - and by combining a detailed material analysis of the instruments with interviews, video and other evidence, I document the practices involved with their use. By rebuilding these instruments, and designing and building other devices, I test my hypotheses through my own practice, and by doing so I refine my results and extend my composition, performance practice and technical design skills to include valuable lessons learned through this research.

The portfolio engages with the three archaeological levels (Listening Situation, Reproduction Stage, Production Environment) and the three areas of the production continuum (Composition, Performance Practice, Technical Design) and through sound installations, crafted media, recorded performances, and the documentation of devices designed for these pieces, it supports the thesis through experimentation and incorporation of results through reflective practice.

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Chapter 1

Introduction

1.1 Electronic Music Performance Practice

The imprudent scholars, who go directly from the universally true to the singular, rupture the interconnections of life. The wise men, however, who attain the eternal truth by the uneven and insecure paths of practice, make a detour, as it is not possible to attain this by a direct road; and the thoughts which these conceive promise to remain useful for a long time, at least insofar as nature permits.

(Vico, 1965, (1744) p. 34) in (Hill, 1988)

The ideas discovered through practice have a great influence not only on the sounds heard when we listen to electronic music, but also on the compositional process, the technology and instrument design and on the practice itself. This thesis examines some of the reflexive relationships and processes connecting these three elements in order to demonstrate the value of documenting, preserving and transmitting electronic music performance practice to current technicians, performers and composers. My practice is directed towards the documenting of such tools, techniques and relationships through experience so as to understand how much and what kind of information is necessary in order to pass on the information to future researchers and practitioners.

My own practice is based on a lifelong interest in machines and electronic instruments. During several years spent producing electronic dance music, releasing two albums and a handful of 12" singles with tracks appearing on dozens of compilations, I developed a live performance practice which included using modular synthesizers, samplers and an early laptop computer and performed in Europe, the USA and Japan.

Popular music studies has built up as a discipline to such an extent that we are now able

to discuss within the academy any kind of music alongside any other. For electronic music studies this is an essential starting point because innovation, both in composition, performance and technical design has always happened across a wide range of musical communities, genres and environments. In this thesis I make no distinction between high and low, serious and popular. Juxtaposition of Stockhausen with King Tubby (Chapters 3 and 4) may at first seem jarring, but the analysis of their approaches towards and uses of particular technologies exhibits a surprising amount of correlation, and underpins the assertion that value is to be found across the widest spectrum of music practice.

1.2 Material Study

In its relatively short history, the practice of electronic music making has changed in many aspects due in large part to the pace of technological development, expanding and shifting into new and predominantly computer related areas as resources have become cheaper and more accessible. The fluid definition of electronic instruments themselves has made documenting practice very difficult, and the fundamental shift away from analogue electronics to the use of computers (often to simulate both acoustic and electronic instruments) can be argued to be as significant as the shift from acoustic to electronic. This boundary between electronic and computer is porous at best, but is significant because the rapidity of this change coupled with the naturally future-oriented scope of electronic music making has meant that some important elements of practice have not been adequately documented before being abandoned or superseded, and much of the early, pioneering research, design, and technique is in danger of being lost altogether. In the case of music made using electronic means it seems that far more energy has been put into pursuit of the new than into fully exploiting existing technologies. To some extent the more efficient and intuitive devices such as channel faders have become so transparent to their users that their characteristics and influences have been almost entirely overlooked. I touch on these influences in Chapter 4.5.1, and such concerns have strongly informed some of my designs including the quadraphonic mixer which is documented in Appendix A.

Very often new technologies are marketed around expanded feature sets, often at the expense of interface quality. Digital Audio Workstations (DAWs) - the current digital versions of the tape recorder - are not sold (or presumably bought) on the merits of the quality of the recording capability, which is taken for granted since it is just the storing of ones and zeros in good time, but by the myriad editing features designed “to fuck around with the audio once it’s been recorded.” (Albini, 2010) Rather than being the primary focus of the device, the actual process and quality of recording has

become almost a by-product. The impediments to efficiency: cleaning the tape path, demagnetising heads, track alignment, tape stock, biasing, splicing, speed selection, waiting for the tape to rewind, etc. have been eliminated, but so have any other qualities of both the materials and the processes of analogue tape recording. Many of these abandoned materials and processes form the basis of my research, as well as the subject investigated creatively through the portfolio works.

The study of electronic instrument practice has parallels to early music studies, but electronic instruments are superseded and considered obsolete at a much faster rate. Documentation in scores is not often comprehensive, and even Stockhausen, who provides photographs, frequency lists etc. often glosses over salient details which remain unrecorded. Knowing which details are significant, predicting which tacit knowledge will disappear or which idiosyncrasies of the instruments are the most influential is almost impossible without allowing for the passage of time. Where possible, I have traced original electronic instruments, and have incorporated these (or reconstructions of these) into my own practice, enabling me to discover which details may have impacted most significantly on the associated historical practices, and what techniques would have been demanded or afforded by the design characteristics of the instruments themselves. Re-creating Stockhausen's W49 filter interface (see Appendix G) enabled me to film Rolf Gehlhaar demonstrating some of the performance techniques that were used in the 1960s. Since Rolf is the last surviving member of the Stockhausen Ensemble from the 1960s, we are very close to losing some of this first-hand information forever. This urgency drives my research. Recreating King Tubby's filter (see Chapter 6.1.5 and Appendix B.1) allowed me to use it in my own practice and test theories of how his studio was configured.

1.3 Case Studies

By focusing on electronic music performance practice in two very different situations in Chapters 3 and 4 and especially the design and use of certain electronic filters both within these contexts and my own practice, I show how a detailed material study of original instruments and practice can contribute to a deeper understanding of the relationships between the various elements and agents surrounding the production of electronic music such as *Mikrophonie I*, *Kurzwellen*, *Spiral* (Stockhausen, 1974, 1969c, 1973). My analysis considers studio practice in the same context as live concert performance practice and this thesis does not make significant differentiation between the two. The scope of this research is limited to electronic instruments designed and made before the early 1970s, and primarily concerns a class of devices called *Stepped Filters* (see appendix F).

The first case-study is based on the use of a high-pass filter - the Altec 9069b - within the context of record production in King Tubby's studio in Kingston Jamaica in the 1970s; the second is based on the use of a band-pass filter - the Maihak W49 HörspielVerzerrer - within the context of performances by the Stockhausen Ensemble in the late 1960s. These case studies show striking similarities of practice and provide the basis for advancing a model of how such instruments influence practice and the sound produced. This provides a useful methodology with which to approach similar subjects, and it documents these practices in such a way as to enhance and inform performance practice for interpretation and analysis of this music.

1.4 Portfolio

My accompanying portfolio is representative of my engagement with what I regard as the *continuum* of music practice ranging from composition through performance practice to instrument design and construction. Whilst this may seem unfocused, I have found it necessary to develop skills across the spectrum in order to be able to make the music I want to make. The extension of my practice into the technical domain has been absolutely vital in enabling me to analyse, criticise and understand the practices which appear in the case studies (Chapters 3 and 4) and is therefore of core importance to the thesis. The ability to identify the sometimes subtle effects of particular technical design parameters enables me to adjust them, allowing me to produce works in the way that I want them, and this is expressed in the accounts of individual synthesizer modules (see Appendix B) and the quadraphonic mixer (see Appendix A), designed and built as part of the portfolio; each design having been necessary for the realisation of a particular portfolio piece.

Since my own music is often process based¹, I also see the technical design side of my practice as an integral part of my musicking (Small, 1998). In *Three Way Conversation* (see Chapter 6.1.1) the composition, if such a thing can be identified in this context, consists as much of the design of the instrument and the characteristics of the recording medium, as it does with the organisation of sound as expressed by the performer, but no element is subservient to any other - hence the title.

The ability to make and mend devices, as well as enabling research into existing practices, also allows me to rescue abandoned machines and to extend the useful lives of many different instruments in my own practice. Crawford describes this engagement thus:

¹I released two albums and several singles under the name *Process* in the 1990s and early 2000s.

Being able to think materially about material goods, hence critically, gives one some independence from the manipulations of marketing [...] Knowing the production narrative, or at least being able to plausibly imagine it, renders the social narrative of the advertisement less potent.

(Crawford, 2009, pp. 17-18)

This awareness enables resistance against the overwhelming consumer culture/technology alignment (Hill, 1988) and aligns with Attali's fourth network, composition, which he presents as an empowering of the individual set against the influence of the market (Attali, 1985). The inevitable improvisation of tools and components involved in instrument design, both electronic and mechanical, can also be aligned with my musical improvisation practice upon which many of the portfolio pieces are based. The failure mode of analogue electronics tends to be far more forgiving, accomodating, and as Cage might say, "interesting" than that of digital equipment² and such failures can (and sometimes must) often be incorporated into my practice. A continuing on and off-stage negotiation with failed and failing machines is a part of my making, listening, and performance practice and is intimately connected with the considerations of noise set out in Chapter 2. The negotiation process strongly aligns my music practice with a craft approach and I do not seek to hide the handmade nature of my instruments or my music; on the contrary, I consider it to be a defining characteristic. Working with hardware allows me a more immediate connection to the materials of electronic sound production and allows me more autonomy of enframement as opposed to working within a more controlled software environment.

1.5 Electronic Designs Skills

In order to understand older technology I have had to use it, and to be able to use it I have had to develop electronics and mechanical engineering skills.³ In many cases, with electronic music instruments, there is no easily defined boundary between instrumental performance practice and electrical engineering practice, and with few resources to employ assistants or maintenance engineers, the electronic music practitioner must have the skills at least to keep his instruments in working order.

With the disappearance of the apprenticeship system that used to operate in studios, and with the retirement of many experienced studio assistants and engineers, older hardware is simply not supported in most institutions, and so without learning these

²which in my experience tends to just stop working and remain recalcitrant until an often lengthy reboot process has "fixed" the fault

³In the case of a repair I carried out on an old EMI TR50 whose perished rubber flexible coupling I managed to fix using an old bicycle inner tube, some vulcanising compound and several clothes pegs, these skills verge on agricultural engineering.

skills myself, this area of my research would not have been possible. The tacit knowledge needed to make music with such devices cannot be learned without practice, but once learned, certain details seem so obvious that it is easy to understand why many have not been documented.

My main instrument is the modular analogue synthesizer which is in a semi-permanent state of flux as an instrument. Modules may be repatched at will, often during performance, but they may also be rearranged, substituted, or customised, allowing the instrument to be completely reconfigured, thus blurring the boundaries between instrument design and performance practice. Building my own modules - e.g. a ring modulator for use in *Testing Testing* (see Appendix B.3 and Chapter 6.1.3) and a varispeed remote for *Electronic Skank* (see Appendix B.2 and Chapter 6.1.5) has extended my practice considerably. This extension of my music making activities through performance practice into design partly explains my reticence to define myself as a composer since I see each element as being as important as the other.

This craft knowledge and experience has been made possible by a natural curiosity to take things apart and find out how they work, allied with the development of a social network that I have built up consisting of my electronic design mentor Graham Hinton, and various other individuals and experts.⁴

It is worth stressing the two separate strands of design skills necessary for building instruments; the electronic and the mechanical. In many cases the electronic design seems the most complicated but in my experience and anecdotally, it is the mechanical design and manufacturing that presents the biggest challenge. To a certain degree, if the logic is right then the circuit should work, and when it doesn't, a few component changes and rewiring here and there will fix it. Manufacturing a control interface without access or funding to make custom extrusions, cast metal objects, precisely punched faceplates, ergonomic plastic handles etc. is much more of a practical challenge, and it is often the interface which lets down many contemporary interpretations, instruments and models.

Some consideration must be given to the feel of the controls even on the component level, when choosing the type of rotary potentiometer for example, so there is a clear overlap between electronic and interface design. In laying out the circuit board, the designer must consider where to situate the various controls so that they will be in the right place for performance, so it seems obvious that an active link between electronics designers and performers is necessary. The composer enters the frame when demanding particular results of the performer who may then require a feature of the interface or circuitry not needed previously. When combining the three roles into one, as in

⁴see Acknowledgements

my own creative practice, there is tremendous liberation in the process of choosing parameters and laying out designs, and one finds that a design decision can be made as a composition decision, and vice versa. Often compositional decisions originate from limitations imposed by technical considerations. The development of my own listening practice and an increased tolerance for noise in its many guises has allowed me to be much more flexible in how my own demands and desires are met within this continuum of practice. This is demonstrated particularly in the Cylinder Pieces (see Chapter 6.1.1).

Repair is a neglected, poorly understood, but all-important aspect of technical craftsmanship. The sociologist Douglas Harper believes that making and repairing form a single whole; he writes of those who do both that they possess the “knowledge that allows them to see beyond the elements of a technique to its overall purpose and coherence. This knowledge is the ‘live intelligence, fallibly attuned to the actual circumstances’ of life. It is the knowledge in which making and fixing are parts of a continuum.” Put simply, it is by fixing things that we often get to understand how they work.

(Sennett, 2008, p.199)

The parallels between making and repairing, design and modification, and composition and improvisation are striking. The path towards building my own synthesizer modules started with modifying existing modules for two main reasons; to enhance the amount of control I had over certain parameters; and to improve or alter sonic characteristics. These modifications might involve substituting one op-amp for another with lower noise and higher bandwidth, substituting one resistor for another of a different value, replacing one timing capacitor with two others and a panel mounted switch to select between them. The design practice overlaps in these situations with performance practice and composition, especially given my own bias towards process-based music practice. Emphasis may change but all three elements are essential components of my own practice, and all three components allow me to gain insight into the practices of others which I document via the case studies in Chapters 3 and 4. Experimentation with components in my own designs has given me insights into component choices and their relevance and influence in the instruments I have been studying.

1.6 A Synergetic Approach

My methodology treats music as a phenomenon occurring as part of a physical system and not as a platonic ideal, drawing from Bateson’s ideas regarding organisms and ecosystems and Fuller’s recognition of the impossibility of understanding an object

without considering its context (Bateson, 1972; Fuller, 1969). This has been echoed by Clarke through his “ecological approach” to listening (Clarke, 2005) and by Waters in discussion of the “performance ecosystem” (Waters, 2007). I will refer to this using Fuller’s terms; synergy or synergetic, relying on Fuller’s definition of synergy:

[the] behavior of whole systems unpredicted by the separately observed behaviors of any of the system’s separate parts or any subassembly of the system’s parts.

(Fuller, 1969, p. 71)

Where possible, I examine some instruments down to the component level and in seeking to understand their precise characteristics I have designed and manufactured reconstructions of some of these instruments (See Appendix B.1 and G). As Fuller implies, a material analysis on its own is of little value unless it is contextualised, so I attempt to show how the material nature of such devices is essential to their identity and use as musical instruments within a larger system. My portfolio addresses the engagement and use of these instruments along the continuum between design, performance and composition practices. The practice element makes it of primary importance that all elements contributing to the production of electronic music, right up to the experience of the listener hearing it (explored in Chapter 2) are considered. The material study does not seek to split the technical sound production from the listening experience; rather it seeks to illuminate causal links and relationships in both directions, and attempts to situate the technology within the wider synergetic framework of the listening environment, particularly with reference to the audible clicks exhibited by the instruments in both case studies.

Instead of regarding electronic devices as ideal models of mathematical processes, or simply referring to circuit diagrams (idealized models of a different nature), I consider them as individual musical instruments. When I need to use ring modulation as an interpreter or as a composer it is therefore important to decide whether I need a passive model based on transformers and germanium diodes, an active model based around a balanced multiplier IC, or a digital multiplier object. The relationship between the chosen device and the rest of the system is of paramount importance, especially in the case of passive electronic devices. These relationships quickly become complex and unpredictable, and are often characterised to a greater or lesser degree by imperfections, most often expressed as noise. Since noise is also an important and often overlooked factor in the “performance ecosystem” (Waters, 2007), Chapter 2 sets out what sort of noise I will be considering and how this sort of noise can communicate information to the listener from any point between composition and audition. This is relevant to both case studies as well as to the portfolio material which explores all of the areas defined

in this chapter.

Sociological aspects of electronic music performance practice must not be ignored, and as Manning suggests, this is necessary for an inquiry into creative influence (Manning, 1999). Although acknowledged, it is beyond the scope of this thesis to address every perspective of the problem. The key areas in which they are of interest are the relationships between members of the Stockhausen Ensemble, the relationships between King Tubby and his protégés, and my own relationships with the extended network of technical experts and musicians⁵ who have helped me to understand the finer points of design in some of these instruments. I would hope that my research could be used in parallel with an in-depth sociological study to shed more light on this subject area and I feel that this is best done by using material research as a measurable foundation on which to base such further work. A study of relationships between composers and technicians along the lines of Born's IRCAM investigation (Born, 1995) would be especially revealing.

My own practice seeks to confront some socio-political problems directly. Ideas of agency, value, noise, progress and, most invidious of all, obsolescence, are interrogated through fixed media, composition, and performance pieces often using found, discarded or second hand materials. *Testing Testing* (Chapter 6.1.3) uses second hand test tone records from car boot sales and charity shops, and a discarded copy of a test tone tape, and is performed using whatever record players are available, including one rescued from a municipal rubbish dump. The oscillators are 1960s Levell types bought for no more than £10 each and the ring modulators are homemade devices. (*Sound of Music*)' (see Chapter 6.1.4) uses a mono vinyl copy of the soundtrack of *The Sound of Music* as its source material. This is an artefact so commonly discarded as to be available for around £1 in almost every charity shop in Britain. These pieces both make music from discarded and abandoned materials. The narrative that lies behind some of my choices, discoveries and inclusions adds an additional personal element to my instruments and although not a central part of my argument, remains important to my own enjoyment of my practice.

Manning frames the creative force in dualistic terms: "one force based upon the nature of the technological resources, the other concerned with the compositional processes themselves" (Manning, 1999). I would like to suggest a third element by considering the performance practice as the interface between these two forces, practice which includes that of the studio realisation by the "technicians." A score is a technical procedure, and in Chapter 3 I present observations regarding King Tubby's studio practice that support the consideration of technical realisations as performances or interpretations. The example which Manning uses himself - Stockhausen's *Elektronische Studie II*

⁵Skills often combined in the same individuals.

(Stockhausen, 1956) - shows the relevance of this third element in a comparison of its original version to the realisation which is included as an example patch bundled with the Max/MSP audio software (Cycling 74, 2011). Whilst Stockhausen's realisation is considered by some to be more an example than an outstanding piece, the Max/MSP version is notable for its sterility. This is an executed technical exercise following the data in the score relating to sine wave frequency, duration, and amplitude; the original version subject to physical processes and manipulation of materials by hand; the Max version achieved through digital algorithms. Although correct, and undoubtedly more accurate than any version made using laboratory oscillators, reverberation rooms, tape editing, and fader manipulation, the removal of the animate and the physical agents from the realisation process as evidenced by the Max/MSP version yields a very different result from the original.

To use Fuller's terms, isolating the "subassembly" of the composition from the electronic instruments and the performance practice changes the behaviour of the whole system. Or to reverse the concept, the composition alone cannot predict or dictate the behaviour of the whole system, and if it can, as in the Max/MSP version, with no noise, imperfections or physical performance elements entering the chain, does it produce interesting results?

By focusing on the technical resources and the performance practice I am trying to find out what kind of mechanisms and details may be involved in electronic music making processes, which have made notable contributions, and which can be usefully absorbed into contemporary practice.

1.7 Accompanying Materials

Throughout the text I refer to three main sources of data which are included in digital format: the Portfolio CD, the Examples CD, and the Portfolio DVD.

The Portfolio CD contains final versions of the portfolio pieces.

The Examples CD contains excerpts of commercial tracks which are referred to in the text but which can only be used for academic purposes.

The Portfolio DVD contains pdfs, videos, audio and software patches organised in folders relating to each portfolio piece. An additional folder, *PhD Materials*, contains miscellaneous materials.

1.7.1 Portfolio CD Track Listing

1. Three Way Conversation (Cylinder Piece 1) (2:21)
2. Ghost Tracks (Cylinder Piece 2) (2:22)
3. Sine Tape Study (3:52)
4. Testing Testing (16:18)
5. Electronic Skank (London) (6:16)
6. Electronic Skank (Edinburgh) (8:12)
7. Spiral (18:32)

1.7.2 Examples CD Track Listing

1. *Short Note With a Few*, Flanger. (Flanger, 1999) (1:29)
2. *Noggin and the Birds*, Oliver Postgate. (Oliver Postgate and Ronnie Stevens, 1963) (0:40)
3. *Original Gonzo*, Mark Dressler and Sean Williams. (unpublished, copyright Dressler, Williams) (1:24)
4. *IPA Skank*, Lee Perry, mixed by King Tubby. (King Tubby, 2004) (0:32)
5. *Andy Warhol*, David Bowie. (David Bowie, 1971) (0:56)
6. *Dub Organiser*, The Upsetters featuring Dillinger, mixed by King Tubby. (King Tubby, 2004) (0:59)
7. *Tubby's Dub Song*, King Tubby. (King Tubby, 2004) (3:09)
8. *IPA Skank*, Lee Perry, mixed by King Tubby. (King Tubby, 2004) (1:05)
9. *A Noisy Place*, John Holt, mixed by King Tubby. (King Tubby, 2004) (0:53)
10. *Mikrophonie I*, Karlheinz Stockhausen. (Karlheinz Stockhausen, 1975) (0:18)
11. *Rebel Dance*. Aggrovators, mixed by King Tubby. (King Tubby, 1989*b*) (0:38)
12. *Beat Down Babylon*, Junior Byles, mixed by Lee Perry. (Junior Byles, 1984) (1:30)
13. *Overture and Preludum - Dixit Dominus*, Rodgers and Hammerstein. (Rodgers And Hammerstein, 1965) (1:08)
14. *The Sound of Music*, Rodgers and Hammerstein. (Rodgers And Hammerstein, 1965) (1:15)

1.7.3 Portfolio DVD File Listing

- PhD Materials
 - / Quad Panner.pdf
 - / Rolf Video.mov
 - / Sean Williams PhD.pdf
 - / W49 datasheet.pdf
 - / W49 Mockup.pdf
- Cylinder
 - / Cylinder Session Photos.pdf
 - / Ghost Tracks.aif
 - / sbkw_open_door
 - / sbkw_door_end.aif
 - / sbkw_door_main.aif
 - / sbkw_opendoor_demo.maxpat
 - / sbkw_opendoor_patch.maxpat
 - / Three Way Conversation.aif
- Electronic Skank
 - / Electronic Skank Edinburgh.aif
 - / Electronic Skank Edinburgh.mov
 - / Electronic Skank London.aif
 - / Electronic Skank London.mov
- Sine Tape Study
 - / sbkw_sines_mix.aif
 - / Sine Sketches.pdf
 - / tape_flute_recording.aif
 - / tape_solo_recording.aif
- Sound of Music
 - / jit.som_sbkw.maxpat
 - / Sound of Music documentation.mov
- Spiral

/ Spiral Portfolio mix.wav

- Testing Testing

/ Testing Excerpt.mov

/ Testing Testing.aiff

Chapter 2

Noise and Fidelity

2.1 An Archaeological Approach

As an archaeologist sifts through piles of earth looking for physical traces of habitation in the form of pot shards, cracked bones, the patterns of chipping on a block of stone, the leavings of ordinary activities, I propose that we can approach the analysis of music practice in a similar way, searching for evidence of human activity through traces left in the audio signal at different levels of remove from the listening experience.

My categorisation is a spatio-temporal division. Working back from the instance and location of the listener’s perception to the initial musical idea, I set out three archaeological levels: the *listening situation*, the *reproduction stage*, and the *production environment*. With the listener at one extreme and the composer or music-maker at the other I offer several strategies by which some of these traces may be identified in each level and be used as the basis for analysis. These levels are partly characterised by the amount of listener or composer control available. With most control available at the production level, the composer’s influence diminishes with each subsequent level, as the listener’s scope for influence and control increases. This is sometimes subverted and examples in the following sections will clarify this.

We can observe parallels between the *production environment*, *reproduction stage*, *listening situation* and the performer, instrument, environment model, aligning this archaeological approach to the consideration of music within a “performance ecosystem” (Waters, 2007). It is interesting to note the inevitable interactions between the three archaeological layers, many of which disrupt the temporal relationships and control patterns exhibited therein.

This chapter outlines a methodology by which noise, primarily in recordings, but also

in live performance, can be treated as a positive component of the audio experience, enhancing and increasing fidelity, value and enjoyment for the listener, and providing valuable information regarding the means of production, use of resources and the composition and performance practice surrounding its creation. An understanding of these kinds of processes provides the raw materials for many of my own music making activities. This is most clearly evident in *Sine and Tape Study* (see Chapter 6.1.2 and Portfolio CD track 3) which magnifies noise elements associated with the tape medium - edit points and tape hiss; ambient noise of the environment; and noise elements associated with the flute. *Testing Testing* uses for its primary material the deviations from the notionally perfect sine wave test tones, and *Three Way Conversation* and *(Sound of Music)'* both include the nature of the media as exhibited through noise as the central elements of composition. Figure 2.1 sets out in pictorial fashion how each portfolio piece relates to each archaeological level and to different aspect of the technical design, performance practice, composition continuum. This is revisited in Chapter 6.

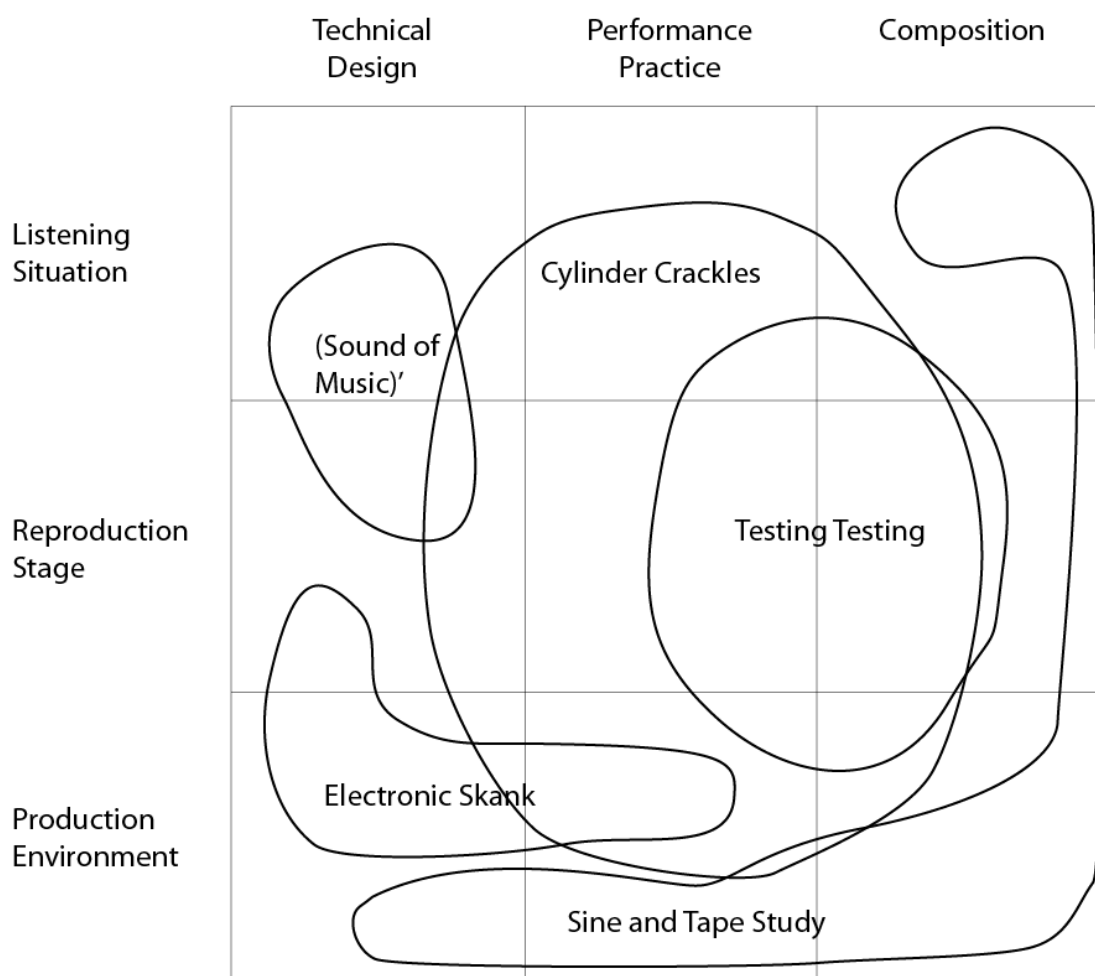


Figure 2.1: Portfolio works in relation to the main ideas explored in the thesis.

Although the sense of noise used in basic communications theory is sometimes useful, the idea that: “To make noise is to interrupt a transmission” (Attali, 1985, p. 36), sets up noise in opposition to a signal and promotes a negative view towards it. Analysing this model in more detail, we can see that far from disrupting communication, noise carries meaning itself, and can also be essential to any meaning that might be carried by music too. Link and Auner tackle the role of noise in recordings primarily from a temporal point of view, invoking nostalgia as a prime motivating factor for the creative use of noise, and analysing the use of what they term obsolete technology mainly for effect (Link, 2001; Auner, 2000). Whilst nostalgia is an obvious effect of the use of older technologies, there are many other more useful mechanisms involved which I set out below.

2.1.1 Noise

Noise definitions have noisy boundaries so my use of the word will inevitably overlap into other areas, but in general I refer to incidental, unplanned, or unwanted sound that arrives at the ears of the listener in addition to what may have been intended by the composer, performer or listener. I am excluding the modernist noises of Varèse and his use of relatively unpitched percussion instruments; futurist (and political) noises of Russolo and Marinetti exemplified by the *intonarumori* (Russolo and Brown, 1986); deliberate and loud noise and distortion as the main compositional element by Merzbow and other so-called Noise artists. It is sometimes difficult to identify whether certain sounds are intended or not, and as we will see below, when incidental noises are deliberately used for effect. For these reasons, my definitions must remain flexible, but examples are given to clarify the different types of noise that will be discussed.

This definition implies an “ecological approach” to listening as outlined by Clarke with reference to what can be *measured* and what is *perceived* (Clarke, 2005). Whilst Clarke focuses on the *perceived*, my archaeological approach often takes a contrary stance by providing a methodology for analysing the *measured*. By breaking this down into layers, it is then possible to trace how these particular elements may carry meaning, or at least provide information and may therefore influence what is *perceived*. An awareness of these mechanisms allows the music maker to develop strategies to exploit them to his advantage, and provides the listener or musicologist increased insight into the music making processes.

In theoretical terms, the early struggle for manufacturers of audio recording and reproduction equipment was to increase the signal to noise ratio, and this was exemplified by the competition between early phonograph and gramophone manufacturers as outlined by Sterne (Sterne, 2003). At some point the signal-to-noise ratio became decoupled from

the rest of the audio experience, being something that could be measured and therefore demonstrably quoted in promotional literature, as opposed to more subjective ideas of sound quality which could only be perceived. This chapter makes an attempt to analyse and understand what price has been paid in the pursuit of this goal; what has been sacrificed, by showing several different examples and alternative approaches to the concept and value of noise, and therefore supporting the act of resistance towards the prevailing consumer driven culture-technology alignment (Hill, 1988).

2.1.2 Fidelity and Authenticity

The concept of fidelity has been used in marketing audio equipment for so long (Sterne, 2003; Milner, 2009) that it has become synonymous with sound reproduction equipment via the word “hi-fi” - an abbreviation of high-fidelity. The famous HMV logo features the dog Nipper listening to a recording which we are asked to believe is clearly recognisable as his dead master’s voice. (The early designs show the gramophone player sitting on a coffin and not a table). The fidelity of the dog to the master, even after death, is used here as an analogy to the faithfulness of the sound reproduction, linking the ideas of devotion and loyalty to the process of listening to a recording of a musical work - the recording as a faithful reproduction of an original event. This is problematic for obvious reasons, principally regarding the process of making the recording and how that original event had to be manipulated in order to make it sound as expected. When multitrack recording and editing became possible, the disconnect between any original event and the recording could be total. In the case of tape music to some degree, and certainly with computer music, it could be argued that no event was actually taking place to be recorded at all. In these more extreme cases, the recording is the event itself.

Are the concepts of fidelity, faithfulness and authenticity freely exchangeable? Is the listener listening to a faithful representation of a composer’s work, a faithful capturing of one particular performance, a faithful attempt by an engineer using technology to translate the vague specifications of an artist into a reproducible format? In order to use the term with clarity the question must first be answered: fidelity to whom or to what?

It is of some value to interrogate some different perspectives on fidelity so that we can understand how the term is being used to justify seemingly opposite trajectories - platonic and phenomenological. Examining noise by means of this archaeological research methodology might allow the f-word to be reclaimed from the home-audio salesmen. The widespread acceptance of the term “lo-fi” as an opposition to squeaky-clean, sanitised audio practices reinforces this position by aligning noise against fidelity.

However, as Frith has observed, these practices are often used to create a sense of authenticity so at the same time (Frith, 1996), this drives a wedge between fidelity and authenticity. Because noise acts as a vector for meaning and value, its elimination can therefore result in lower fidelity. I suggest that fidelity can be used to justify techniques employed in either direction and can therefore be used more constructively to inform analysis of a wide range of music, whether recorded or performed live, noisy or not.

When we use the concepts of fidelity or authenticity we must therefore be careful to acknowledge the subjectivity of the terms. We can speak of authentic 1970s performance practice but that might not in itself be an example of fidelity to the composer's idea of the piece. As I will demonstrate in Chapter 4 the composer's conception can itself change over time, making these concepts even more hazardous to lay claim on.

The idea that a recording, even a live recording, is somehow authentically representative of an event is unsustainable, technologically. And when we consider electronic or tape music the argument collapses completely. What is an LP record of *Kontakte* (Karlheinz Stockhausen, 1960) faithful to? Are we dealing with Platonic ideals of compositions or can we think of these recordings as works in their own right yet? Is the Max/MSP version of *Elektronische Studie II* (Stockhausen, 1956) more faithful to the idea of the composition than Stockhausen's own realisation of it? The noisy, human, physical element is the big distinction here, and acts in opposition to the prevailing marketing-led high-fidelity grammar.

I must also deal with questions of authenticity in my own practice. Whilst not being necessary for interpretation it can be very useful to know the details of technology and practice so that they can be employed or ignored deliberately. Previous developments can be built on or contested only if they are known about and understood. Without this there is no trajectory and only randomly directed practice, with a high risk of valuable techniques being lost, and worse, much greater likelihood of practice being driven by the market - by corporate-culture/technology alignment rather than creative-culture/technology alignment. By breaking down music into three archaeological layers and trying to identify component features, gathering clues from the noisy traces of instruments, technologies, performers and media, it is possible to gain significant insights into historical practices which can then be assessed and used or discarded at will.

The following sections examine each of the three layers and outline the kinds of noises at risk by acceptance of the low-noise approach to fidelity, and what sort of meanings or information they might be carrying.

2.2 Listening Situation

2.2.1 Listening Practice

Cage's famous anecdote about hearing two sounds, high and low, in a visit to an anechoic chamber (Cage, 1968) illustrates the practical impossibility of a total absence of noise. Cage relates an engineer identifying the low frequency sound as the sound of the listener's blood circulation system, and the higher frequency sound as the nervous system. We can work back from here, outside the listener's body, to consider the sounds present in the listening environment such as central heating drone, fridge motors, passing traffic, transformer hum, and a host of other sounds. It becomes obvious that there are no conditions in which music can be listened to without some sort of noise component. This is where listening *practice* (Sterne, 2003) can be emphasised, and although perhaps not the first to promote such an idea, Brian Eno's deliberate use of this phenomenon confronts this noise problem by absorbing it into his compositional strategy:

In January of this year I had an accident. I was not seriously hurt, but I was confined to a bed in a stiff and static position. My friend Judy Nylon visited me and brought me a record of 18th century harp music. After she had gone, and with some considerable difficulty, I put on the record. Having laid down, I realized that the amplifier was set at an extremely low level, and that one channel of the stereo had failed completely. Since I hadn't the energy to get up and improve matters, the record played almost inaudibly. This presented what was for me a new way of hearing music - as part of the ambience of the environment just as the colour of the light and the sound of the rain were parts of that ambience. It is for this reason that I suggest listening to the piece at comparatively low levels, even to the extent that it frequently falls below the threshold of audibility.

(Brian Eno, 1975, *Discreet Music*)

Unlike Cage's *4'33"* (Cage, 1960), this does not rely on concert hall listening practice, but rather, proposes a new listening strategy, opening up the possibility of considering the inclusion of domestic sounds within the listening ecosystem and actively validating a home-listening practice. Here, Eno is deliberately allowing the *listening situation* sounds to form part of his composition and is acknowledging and including the affordance of such noises to change each realisation of his composition. Through the sleeve notes he is laying claim to the sounds of the environment as part of his composition. As a record producer, he has also made this music deliberately to be listened to at home. It is not a recording of a live event or concert, and if the listener follows the suggestion of the sleeve notes, a higher fidelity to the composer's intentions

is gained by admitting the “ambient” *listening situation* sounds into the listening experience. Reducing noise and maximising signal-to-noise ratio in this example clearly leads to a low fidelity experience of *Discreet Music* with respect to the composer’s intentions. Instead of ignoring them or expecting an ideal *listening situation*, both Eno and Cage acknowledge and lay claim to the sounds in the *listening situation* and allow them to inform their composition.

In *Discreet Music* we have, printed on the record sleeve, the intentions of the composer. This is uncommon, although shorter instructions such as “Play Loud” occasionally appear on other records or CDs. It is interesting then because it proposes and endorses a mode of listening that goes against the assumed best practice. The moment of revelation for Eno himself is described in the notes, but once this has been realised by the listener through experience, it then becomes much easier to apply this technique to other situations, and hence to more easily question the assumed best practice.

The idea of fidelity is ultimately linked to some notion of an original, sanctioned, authorised or ideal listening experience, and, we can assume, one set out by the composer, creator or music maker. But how do we know what was intended in the absence of sleeve notes, programme notes, research or context? The above example is slightly disingenuous in that it is rare that we are instructed on how best to listen to a record or piece of music, but what it shows is the possibility of using a very different listening practice to legitimately experience music as the composer intended.

Once we accept Eno’s very quiet listening practice as a way of engendering a high-fidelity listening experience, we may choose to apply the technique to other records or indeed, to other *listening situations*. The assumed hierarchy of fidelity can then be interrogated, and with it, the notion of the original or intended musical experience. In the absence of any instructions or advice from the composer do we use the listening practice associated with the hi-fi shop or do we use Eno’s quiet listening technique? Without guidance, what exactly are we being faithful to? If Eno’s technique works, as it did for him with *Discreet Music* and with the harp music, following the recommendations of the audio equipment manufacturers and salesmen is therefore not the only path to fidelity, and can sometimes lead us away from it. Surely we are free to use any technique we like in its pursuit, and we also have the ultimate choice about what we are trying to be faithful towards. Whether we choose to listen to Beatles singles on a Dansette, Einsturzende Neubauten on an iPod whilst travelling on the train, or dub reggae on a large sound system, if those are deliberate choices then they are equally valid ways of pursuing a high fidelity listening experience.

Because fidelity is so connected with experience, that makes it time dependent - i.e. the passage of time leads to changing concepts of fidelity even concerning the same

piece of music. Personal associations that the listener has with particular tracks might strongly influence how she feels they should be listened to in the future. The fidelity might be towards remembering or recreating the original *listening situation*, something necessarily quite removed from the composer's intentions.

It seems that an objective sense of fidelity is hard to define and agree on. Even where we get instructions or notes from a music maker, we must accept that they may have reconsidered and changed their minds since those listening instructions were given. This would seem to preclude the possibility of pursuing fidelity in the objective sense, so rather than abandoning the term altogether, perhaps a more dynamic and subjective definition may serve us better. Arguments may be made about the efficacies of different listening practices, and the concept of authenticity may be used effectively to support such arguments.

Milner's reports of tone tests (Milner, 2009) are good examples of how the salesmen have sought to claim the *listening situation* as their own, and have sought to peddle one particular listening practice and to ally this with the notion of fidelity throughout the history of home audio. Link argues that noise itself contributes a different sense of authenticity to the recording in providing evidence of its establishment in the past. He concludes that "Noise thus emancipates the perception of authenticity from the authority of any original." (Link, 2001, p.38)

Of course, listeners may make up their own minds about how to listen to each recording, and they may freely choose whether or not they are pursuing fidelity, using whichever notion of fidelity they please. If we can accept that one common notion of fidelity is related mainly to expensive audio equipment, the latest technologies, and consumer culture, but that this is not the only option, then we can look deeper into the next levels and start to reassess the role of noises and imperfections exhibited therein.

2.2.2 Suspension of Disbelief

Link observes that listeners have developed strategies in order to "Hear through" noise (Link, 2001, p. 35). Coleridge's concept of the "willing suspension of disbelief" (Coleridge, 1817), more often used in reference to theatre, is apposite, at least in its application to the recording as a fictional representation of a single performance. Of course, not all recordings attempt to convince the listener that they are such representations. The countless harmony vocal tracks recorded by Freddie Mercury on *Killer Queen* (Queen, 1974) or *Bohemian Rhapsody* (Queen, 1975), or Jimmy Page playing guitar solos accompanying himself on rhythm guitar at the same time, tacitly acknowledge the fact that the recordings are constructions, but suspension of disbelief

on the part of the listener in no way diminishes the impact or experience of listening to the music.

The rituals involved with perhaps dimming the lights, inserting the CD and sitting comfortably on the sofa in between the speakers are not so different from settling into your seat in the stalls, waiting for the house lights to dim, the curtain to be opened and the actors to appear on stage. The focus of attention is keenly directed towards the record or the play, and the contribution of the environment, central heating hum, people rustling their programmes and settling themselves into their seats, are not necessarily negative. They may actually contribute to the sense of spectacle.

2.2.3 Masking

Noise acts through the principle known as masking to cover or to reduce the audibility of quiet sounds. If noise is filtered so that the appropriate frequency range is selected, it is possible to reduce the effects of intermittent or persistent sounds which may otherwise cause interruption or irritation. This technique is often used by acoustic designers such as Arup Acoustics to mitigate against sonic distractions in open plan offices. Here, the quality of noise is of great importance. In our *listening situation*, an absence of noise, such as can be experienced in an anechoic chamber, means also an absence of the masking effect, and this can reveal all sorts of previously unnoticed sources of noise and distraction, many which might then become intrusive and annoying.

In *Discreet Music*, Eno's instruction to listen at low volume allows the possibility for external, environmental sounds of the *listening situation* to actively mask the music, with the intention of provoking or helping the listener to hear such environmental sounds in a musical context, or even as music. Although this is an extreme example, being aware of *listening situation* sounds is a sensible strategy for the music maker. This may be taken to the extreme in relation to the current trend for mastering music extremely loudly, with a small difference between peak and rms values.¹ Indeed, it has been suggested that the tiny dynamic range and loud volumes commonly associated with young urban mp3-player users is an indication that music is being used to mask environmental noise. At the other end of the scale, the full dynamic range available with 24 bit recording is only of theoretical value as, relative to a bearable maximum level, the lowest level signals will always be masked to some extent no matter what the *listening situation* is.

The amount and quality of noise that is accepted in the live situation as opposed to the home is enormous. As a sound recordist I have made many recordings of live concerts

¹See Milner's discussion of the so-called "Loudness War" (Milner, 2009)

and they invariably feature audible page turns, coughing, creaking chairs, the occasional split note or straggled entrance, making very few of these recordings acceptable for commercial release, however, many have been fantastic performances. One particular hall has a dreadful central heating noise problem which has had a negative impact on a number of recordings, but which has gone unnoticed by many performers not familiar with listening practices associated with recording. The listening practice associated with live music, both as performer and audience, thus seems to be more forgiving than that associated with recordings.

2.3 Reproduction Stage

2.3.1 Medium Quality

Progress in recording media is normally associated with better quality sound, however, examples such as the compact cassette and the mp3 have prioritised portability over sound quality and this compromise is exhibited in reduced frequency response and their own characteristic noises. The concept of better quality in this context is not quite as obvious or as easy to define as expected. Aesthetics is a factor in this, which gives more critical weight to Clarke's *perceived* approach over the *measured* approach, but it is worth investigating whether measurable qualities can tell us anything useful.

Whether we hear tape hiss, rumble, wow and flutter, vinyl crackles, clicks and scratches or mp3 compression artefacts, this is all being introduced after the composer has relinquished control over the master recording, but such sounds are not quite as individualised as the higher level *listening situation* sounds. The fundamental tape compression artefacts or vinyl surface noise will initially be the same for each manufactured copy but each copy will, to a greater or lesser extent, develop its own individual noise characteristics subject to the listener's treatment of the media or simply as a result of natural ageing. This implies that once defined, the listening experience is not necessarily static, and as the reproduction medium changes its characteristics, or is replaced, re-recorded, remastered, reformatted, the listening experience changes with it. This challenges Attali's conception of the characteristics of his third network where he asserts that "replication is always imperfect and doesn't create anything new" (Attali, 1985, p. 33). It is the very imperfection which creates something new, and is exactly what I extend and magnify in my portfolio.

2.3.2 Active Engagement

My portfolio piece (*Sound of Music*)’ takes this idea as its subject matter and explores the departure of one record from its originally encoded musical content through the medium of noise by separating the noise from the music and presenting just the noise (see Chapter 6.1.4).

(Please listen to Example CD track 1, Flanger: *Short Note with a Few*)

But *reproduction stage* noise can be a critically important consideration for the music maker. Flanger’s *Short Note With A Few* from the album *Templates* (Flanger, 1999) starts from silence with an increasing number of clicks and crackles that gradually seem to form themselves into some kind of rhythm. This noise-based rhythm is then seamlessly transformed into the sound of an acoustic drum kit. The transitions from media noise to electronic rhythm and from electronics to acoustic drums are handled in such a way as to make it impossible to perceive any single transition point from one to the other. Although I have not carried out these tests under carefully controlled conditions, I have played this record to people on many occasions, usually within the context of listening to lots of records at home, and occasionally in seminars or even at paid gigs, and it is always entertaining to observe at what point different people perceive the music to have started. This piece far more than any other has such a porous boundary between its identity as a piece of music and its presence on a recording medium that it’s beginning and the identification of its beginning actively challenges the listener’s perception and judgement of this relationship. By the end of the track, which is the last track on one side of one of the records, the listener is less sure that the end of the record has been reached, and if so, whether the sound of the needle running into the run-out groove is part of the music, part of something added to the music at the *production stage* or something unique to that particular pressing. I also have a CD release of the album and the impact is entirely different as there is no argument about when the track starts - the ID point is fixed and the time display gives you the visual feedback of exactly how many seconds have passed, framing your listening experience within its harsh and unequivocal digital limitations.

2.3.3 Clicks and Crackles

Whilst the above example remains in the realm of the *perceived*, we can make useful inferences from measurements associated with some of the noises we hear. Even listening to a digital sound file, if we measure the frequency of clicks on a recording at 1.3 clicks per second for example, we can make a pretty good guess that at some point in its history it had existed as a 78 rpm disk. 2.6 clicks per second implies a 160 rpm

Edison Cylinder recording. A scratched CD might exhibit clicks at a rate of between 500 and 200 per minute depending on how far away from the centre of the disc the track is located. Repeated clicks at other frequencies could point to it having been pressed as a 33 rpm or 45 rpm record. This may reveal some social information such as whether it was a 7" single or part of an album and therefore whether it was intended for a teenage or adult market, whether it was written to stand on its own or to be part of a larger musical work.

This difference also implies a difference in listening history: a 33 would have been listened to at home and not from a jukebox for example, but both scenarios imply all sorts of social precursors such as a certain amount of confidence in quality or perhaps a sufficient level of resources enabling the music to be pressed onto vinyl rather than distributed as an mp3. To stretch the idea further, some connection can be made to artists such as Led Zeppelin and post-Syd Barrett Pink Floyd who made a point of not releasing 45 rpm singles, preferring instead the album format to carry their musical output as a whole. Data obtained by measuring this kind of noise could help support or refute such an assertion². As with most of the information revealed through these different methods, they do not necessarily reveal anything of great moment if taken in isolation, but combined with sociological, musicological, anthropological research, (i.e. synergetically) they can be of great benefit.

Such audible cues can also communicate temporal information. If we hear evidence of 33 rpm, we know that it is likely that our recording was made available on vinyl between 1955 and the present day. An Edison cylinder recording might lead us to conclude that it was probably recorded between 1882 and 1920. Of course, recognising that such audible cues have an effect on the listener enables the artist to experiment with media format choices deliberately. In my portfolio I include documentation of two original pieces made using Edison cylinders as the recording medium. They were made in 2011 and undermine any assumption about the temporal information encoded in the noise of the recordings, however, this expectation of age is an integral part of the piece, and the noise of the cylinder is also an active part of the music itself alongside the sound of a homemade electronic "infra-instrument" (Waters, 2007) made in 2010 based on a STEIM Cracklebox (see chapter 6.1.1). Active use of techniques to emulate the effects of different media should be considered as part of the third and final layer, the *production environment*, but it must be acknowledged that the capability to simulate or fake all of these characteristics exists, and caution must be used when relying on data derived from these methods.

Mp3 compression artefacts can tell us something about the way in which the recording

²traces of 45 rpm crackles in a Led Zeppelin recording could indicate an existence at some point of that track on a 7" or 12" single.

is being used or is intended to be used - it is unlikely to be used in critical listening circumstances or to be played on a sound system for example, but it is likely to be available to a large number of people via the internet. It is perhaps more likely to be or to have been listened to on the bus or on a train, along with the attendant noises of such mobile listening.

It is possible to conceive of a listening history, with surface noise being the evidence of dust - dead skin being literally the physical remains of previous listeners - or scratches due to careless handling. My sound installation piece (*Sound of Music*)³ focuses on this noise as its subject (see Chapter 6.1.4). These kinds of sounds can be interpreted as evidence of physical interaction after reproduction. The composer can have an indirect influence here by choosing the original medium of reproduction and mastering for that, however, the music will invariably end up being copied to tape, burnt to CD, or ripped to mp3 and will assume the additional noise characteristics associated with each format on its journey towards the listener.

One of my favourite records is a red vinyl 7" copy of Oliver Postgate's *Noggin and the Birds* from 1963 (Oliver Postgate and Ronnie Stevens, 1963). This is an audio version of an episode of the children's television show *Noggin the Nog*, and the picture sleeve featuring Peter Firmin's artwork. This, coupled with the lovely bright red colour, and the 45 years that the record has existed have all combined to ensure that the record is so scratched that it is almost unplayable. Hearing the scratches though, for me, is an integral part of the nature of this record as a thing and not as some sort of transcription of an ideal work. It feels natural too - if the producers were worried about small fingers damaging the vinyl and had longevity as a priority they probably wouldn't have pressed it on bright red vinyl. It is asking to be played with as a pretty toy, so the scratches that I hear today are almost pre-programmed into it by virtue of the design choices involved in its production.

(Please listen to Example CD track 2, Olver Postgate: *Noggin and the Birds*)

Analysis of clicks and crackles relies also on a listening practice. In a parallel to a traditional archaeologist's experience informing her differentiation between a pottery shard and a piece of stone, the sound archaeologist must be able to hear the significant detail in noises such as clicks and crackles to be able to weigh their significance. This is an ongoing learning process and relates directly in my own research to an analysis of the clicks on two vinyl issues of the same recording of *Mikrophonie I* (Karlheinz Stockhausen, 1966) (Karlheinz Stockhausen, 1975). The clicks made by the W49 filter instrument can easily be heard as natural vinyl artefacts, and it is only by comparing the two records that the original of some of these clicks can be properly identified.³

³Clicks which appear in the same place on both records emanate from the instrument and clicks

2.4 Production Environment

2.4.1 Sound Quality versus Performance Quality

At the production stage decisions are taken by artists, composers, instrumentalists, engineers, producers and others about what sounds are to be included on the studio master recording. Many of these decisions are likely to be trade-offs between variables such as accuracy of performance versus feel, timing versus groove, distortion versus clarity of intention, etc.. Expectations relating to sound quality can have a great effect on the approach used in the studio, and evidence of varied practices can be found in accounts from engineers, producers, and artists. A common thread in such interviews is the struggle to convince record company executives or A&R men that mistakes or noises are an acceptable price to pay for including the best take in terms of performance.

Bob Johnston's⁴ account of his explanation to Bob Dylan about accepting these errors, and the subsequent value of how the band recovered from error after error in the recording of *Rainy Day Women #12 and 35* (Bob Dylan, 1966) frames the decision in terms of the artist standing up to the both the band members, session musicians and record label (Johnston, 2010). The implication here is that these other elements can also exert strong influence on the noise or error content of the final master recording, reinforcing the obvious point that music making is rarely the result of one person's work.

Milner relates how Bruce Springsteen's album *Nebraska* (Bruce Springsteen, 1982) was recorded on a four track Portastudio to cassette which was then left in Springsteen's denim jacket pocket for a few months before finally being used as the master for the album. It sounds like a story deliberately concocted to lend an air of authenticity to the record, to imbue it with a sense of honest DIY blue-collar hard graft, but if we accept this account as the truth, something unique was captured in those original recordings that was deemed to be so good that it was worth accepting the reduced sound quality, much greater background noise and variability instead of rerecording the songs in a professional studio. Even the mastering engineers, Bob Ludwig and Dennis King, struggled to be able to master it because of its quality (Milner, 2009), however, it still reached No. 3 in the Billboard charts and was the Rolling Stone Critics Choice as album of the year in 1982.(White, 2011)

Although both these examples can be read as exceptions to common studio practice, nonetheless they highlight some of the forces working in opposition to the composer/-

which are only present on one record are artefacts of the recording medium.

⁴Producer of Bob Dylan, Johnny Cash, Leonard Cohen, Simon and Garfunkel, and boss of CBS in Nashville.

music maker, at the same time as showing how the trade-off between sound quality and performance quality can favour the latter even in extreme circumstances. There is no indication that Springsteen favoured the 4-track cassette master for its “lo-fi” sound, indeed, it is not until the 1990s that “lo-fi” becomes a recognised production strategy.

Mistakes in the performance, although rare in contemporary recordings, mostly due to the ease of error correction and editing in digital workstations, can be thought of in terms of a transmission noise - the corruption of the intended message. Timothy Day recounts two examples of this more relaxed attitude in the classical recording environment:

Anne-Sophie Mütter very much disliked cutting and would rather ‘keep the string that doesn’t speak or other minor imperfections than lose the spirit. Reality isn’t perfect.’ André Previn believed in making ‘very, very long takes, a movement at a time,’ and only redoing a section if someone knocked over the cymbals, not for the sake of a split semiquaver, which he was perfectly willing to let go, for which reason, he thought, the London Symphony Orchestra liked to record him. (Day, 2000, p.27)

One common factor in all these examples could be that mistakes are only acceptable once the artist has proven beyond doubt, commercially as well as critically, that they have achieved a certain status so that perhaps there is an imbued confidence that these noises are deliberate and not the result of cost cutting or incompetence.

2.4.2 Hum

If we go back to the studio we may find amplifier hum - if measured it is likely to have a fundamental frequency of 50Hz or 60Hz, suggesting geographical location (North America mains is 60Hz, European 50Hz) - or if it is a different frequency it suggests that the entire recording may have been speeded up or slowed down at some point. In the WDR Studio für Elektronische Musik, the early method of varying the speed of the tape recorders was to vary the mains frequency of the studio! Volker Müller, who worked at the studio from 1970 until 2008 demonstrated how this was done in a visit in August 2011. At EMI’s Abbey Road Studios, the engineers devised a way of varying the tape speed by using a battery powered transistor based oscillator (a Levell TG150DM) which featured in most Beatles recording sessions and was actively adjusted by many different people during recordings and mixdowns (Ryan and Kehew, 2006)⁵.

Accurate measurement of any hum present in recordings allied with an understanding

⁵See also Chapter 6.1.3

of the techniques and technologies used in the recording processes associated with the relevant studio may possibly yield evidence of studio sound manipulation. The noise could in this way, directly carry information about studio performance. 100 Hz hum coupled with a strange timbre could imply a passage recorded at half speed and later doubled - a famous example being George Martin's speeded up piano playing on The Beatles *In My Life* (Ryan and Kehew, 2006). Whilst this remains a possibility, it is very remote given the high quality of such recordings and the significant problems of access to original multitrack masters and techniques for accurate analysis. Nevertheless, it is the kind of possibility that could be explored by a researcher seeking traces of such studio based performance practice.

Hum is a useful marker for archiving and tape restoration professionals often revealing the original tape speed of the recordings by matching it back to 50 or 60 Hz (Hess, 2012).

Such a hum may imply being in the presence of the potential for high volume levels, a low production budget or even a decision to trade a brilliant performance or take against a lesser performance with a cleaner audio signal.

2.4.3 Nostalgia

There are many types of information that are deliberately encoded in the *production environment* that engage with the semiotic potential of the other levels. The availability and use of noise adders such as the PSP Vintage Warmer (PSP Audioware) or Vinylizer type plug-ins which emulate the effects of tape compression and distortion, and vinyl surface noise is commercial acknowledgement of this semiotic potential of certain reproduction technologies. Frith analyses such techniques as used by Portishead, implying this to be a way of creating a false sense of fidelity by association with something old and therefore indisputably authentic, even when the technique is deliberately revealed when the background noise is cut to digital silence at certain key moments (Frith, 1996). The nostalgia factor is also cited by both Link (Link, 2001) and Auner (Auner, 2000) to great effect to describe certain production practices, and this argument is so convincing that there is a danger in accepting this as a blanket assumption; that the use or emulation of older technology necessarily implies nostalgia and is used to fake authenticity.

There is a parallel in music production technology concerning the reasons behind the use of vintage studio equipment from the 1960s and 1970s. Whilst some engineers and producers might use vintage microphones and techniques to recreate the sound of a bygone era as Frith points out (Frith, 1996), especially effective in the drum sound on Amy Winehouse's *Back to Black* (Amy Winehouse, 2007), engineered by the

Dap Kings (Tingen, 2007), many others will use such equipment to get the very best sound possible. Although nostalgia might be one of the reasons, there are other more pragmatic reasons for not using the latest technology, and it is dangerous to align the use of older technology exclusively with nostalgia as this plays into the hands of the commercially driven technology narrative of dynamic obsolescence (Hill, 1988).

An explanation that takes account of the material nature of electronic instruments may find that analogue technology reached its peak in the 1970s and some of the older designs, especially using hand-made components, have never been bettered. Writing about the sound quality of recordings made at Columbia's studios, Milner quotes producer Tony Bongiovi in support of the idea that recording technology was better in the 1950s (Milner, 2009, p. 49). Newer technologies have used more mechanised labour and surface-mount components, and are cheaper to manufacture, but these are cost-cutting strategies which often lead to an inferior sound quality. The gigantic budgets of the German broadcasters in the 1950s and 1960s allowed companies such as Neumann, Maihak, Siemens, Tab, and Telefunken to produce studio equipment of the highest quality where price was no object. With the waning of support from large broadcast corporations and the burgeoning home studio consumer market, equipment manufacturers now find themselves in a different commercial situation and more able to get away with selling low quality gear with built-in obsolescence.

2.4.4 Tape Hiss

During my PhD studies I set up The Tape Rooms recording studio in Bristol with a colleague, the main priority being to rely exclusively on analogue technology. For multitrack recording we used a Studer A80 Mk I 2" 16-track tape machine; for delay, automatic double tracking (ADT), flanging and phasing we used a Studer A80RC 1/4" stereo tape machine; and we mixed down to another Studer A80RC 1/4" stereo machine.

(Please listen to Example CD track 3, Dressler and Williams: *Original Gonzo*)

The most striking experience during the two years the studio was running was the very first session in which we recorded some electric guitar and saxophone to two-inch tape⁶. Having done a lot of recording mainly using a digital workstation on computers, my colleague and I were both absolutely amazed by the way the instruments sounded when playing back the tape. The guitar sounded immediately different to any digitally recorded guitar either of us had heard before. It had no surface noise or other coloration that vinyl reproduction would give it, but whether it was the slight tape hiss, the

⁶The version on the CD has additional overdubbed drums in the production style used for Syd Barrett's *The Madcap Laughs*, but all is based around a short guitar and sax jam.

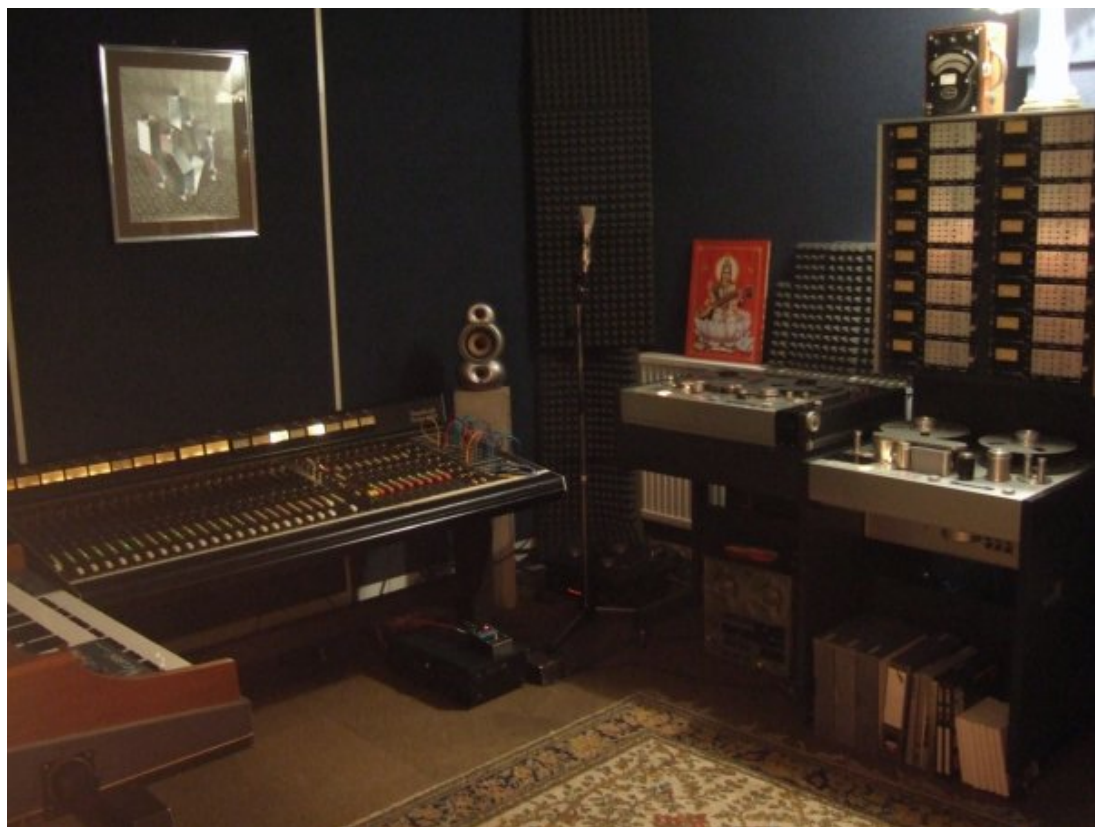


Figure 2.2: The Tape Rooms. L-R: Welson organ, Soundcraft 1600 mixing desk, Studer A80RC stereo recorder, Studer A80VU Mk I 16-track master recorder

amplifier hum or the distortion and compression imparted by the transduction of the sound waves onto magnetic tape, the sound was somehow in the same category as a Led Zeppelin or PJ Harvey *Stories From the City, Stories from the Sea* (PJ Harvey, 2000) guitar sound - it sounded like a record. It had the sound quality at least, of something you would want to go out and buy as opposed to being just data. As Milner points out, this is not a claim for accuracy in the transcription (Milner, 2009, p. 228), but more an aesthetic judgement shaped by a lifetime's listening practice.

Admittedly we were listening in a specialist manner to this recording, and it is impossible to untangle the experience from the emotional situation and the sense of achievement I had after spending several days troubleshooting the Studer tape machine. More aligned to Clarke's *perceived* rather than *measured* listening (Clarke, 2005), the framing noises in the control room - the motors spooling, the tape sliding past the heads and the needles flicking back and forth inside the VU meters - may have added their own elements to our studio *listening situation* experience, and there is no doubt that nostalgia was a factor, but there are many other reasons for using such technology, not least for the sheer aesthetic pleasure of it.

In the same way that Mütter and Previn could live with some less than perfect playing whilst capturing an overall excellent performance, the inevitable tape hiss encountered with this method of recording, although not nearly as noticeable as one might expect, is not added deliberately, but is perfectly acceptable within the context of creating a sound in the recordings with the desired sort of quality.

Although working on a DAW to record and mix Johnny Cash's posthumous *American VI* (Johnny Cash, 2010) album between 2003 and 2006, David Ferguson actually added recordings of tape hiss on most of the songs primarily to cover up edit points (Tingen, 2010). This is a practical example of the masking effect (see section 2.2.3) being employed constructively, but the choice of using a recording of tape hiss rather than any other source of noise is perhaps relevant to the fidelity or authenticity question. In classical recording (and film sound) it is common practice to record some room tone by recording the studio or hall with exactly the same microphone setup as the main recording, only with nobody there, so that this material may be used to bridge any spaces between movements or to cover any exposed edit points. Ferguson's use of tape hiss seems like a direct equivalent but instead of using the authentic sound of the absence of musicians in space he chooses the absence of music on tape. Tape hiss came to my rescue in the mixing of my recording of *Spiral* (see Chapter 5.6.2)

2.4.5 Resonating Bodies

Anyone who has tried to record a grand piano with microphones positioned close to the strings and hammers will have been faced with decisions about how to deal with the sound of the dampers being released from all of the strings as the sustain pedal is pressed. This sound, with which the pianist is familiar, is not expected by the listener, but it is part of the sound of a piano being played, and making production decisions on how to deal with such a sound directly tackles the question of fidelity. Should this sound be removed in order to create a sound that the audience feels more comfortable with? Has the authentic sound of a piano become that of a recording of a piano in the same way that the sound of a Hollywood punch, ricocheting gun-shot or squealing car-tyre have come to represent events despite their disconnection from what such events actually might sound like? Is this therefore fidelity to the ideal of a piano, the history of piano recording, or an actual piano in a space? Again, there is room for interpretation and therefore the possibility of varying and contradictory presentations of high fidelity piano recordings.

The human body is a source of generally unwanted or unmusical noise and its interaction with the environment. Sounds such as page turns, creaking chairs, breathing, coughing can contribute to a sense of presence of performers in a recording which can sometimes

be useful. From the overt use of a looped cough which starts Black Sabbath's *Sweet Leaf* (Black Sabbath, 1971) to the sound of Clyde Otis kicking over a chair in the middle of Dinah Washington's *What a Diff'rence a Day Makes* (Dinah Washington, 1959) (Johnston, 2010)⁷, such sounds reveal the presence of people and can be heard as the extra-musical traces of the music makers. This may be of no interest or concern to most listeners but can act to bring the listener closer to the process behind the making of the record and can conjure some of the atmosphere of this environment. Indeed the listener may not notice the sound of the chair being kicked over, and it would certainly be difficult to identify it as such without the information from the interview, but it is therefore a good example of what can be discovered by combining noise analysis with other research methods.

2.4.6 Off-mic Mutterings

Sounds made by singers picked up before the start of songs, in between verses, and during the songs themselves, form a class of their own. Pete Waterman describes the impact generated by one of the Beatles (Paul or John) directing the "One, two, three, four" preceding their opening tune towards the audience rather than off-mic to the rest of the band. The count-in prepared the audience as well as the band and Waterman describes the resultant positive connection between band and audience in colourful terms (Waterman, 2009). The count-in is effectively a technical part of the production process, but in its simplicity it easily communicates something about the music which immediately follows it that is accessible by all listeners.

In live performances or recordings of live performances the preambles or spoken introductions to songs are almost expected, and in such realtime experiences shared between artists and audiences they can often be conversational and can offer immediate enhancement or explanation of what is about to be played and heard. The spoken interludes between tracks on the Woodstock album (Various Artists, 1970), labelled as "Stage Announcements" on the sleeve notes, contextualise the music on the disks by referring to the weather, the people, the political climate, the drugs, and in their noisy, un-musical nature, positively enhance the effect of the music, giving a great sense of authenticity to the event.

The inclusion of mutterings, commentaries, introductory remarks and other spoken segments on a studio album is a much more calculated technique of production. To assume that it is honestly capturing the event or events surrounding the recording process in a similar way to that of a live album is risky. There will always be some extraneous signals recorded at the start and the end of a take, but these are usually

⁷You can hear it about 25 seconds into the track.

removed by fader manipulation, automation, or editing when mixing. Circumstances in Jamaica in the 1970s allowed the recording process to be separated from the mixing process to the extent that one multitrack recording might be mixed and remixed many times, often by different people. These multitrack recordings would still have had the noisy, unwanted, unmusical off-mic mutterings before the start and after the end of the desired music content.

(Please listen to Example CD track 4, Lee Perry: *IPA Skank*)

Two notable producers and practitioners of Dub, Lee “Scratch” Perry and King Tubby, used to using noisy elements such as banging stones together, dropping spring reverb units, emulating cows mooing, and much more, quickly saw the creative potential in much of this spoken material. Many of their mixes are introduced with these spoken passages, sometimes fragmented and often swathed in echo and reverberation so that sometimes the words are unintelligible and the content becomes another noisy musical element of the track.⁸ Dub uses many different types of noise, both consciously and through serendipity. These are discussed in more detail in Chapter 3.

Dub interacts with all three archaeological levels in a number of different ways. The *production environment* relies on the use of pre-recorded material (*reproduction stage*) as its source, often mixing live to an acetate disc - incorporating the *reproduction stage* into the *production environment* - that is to be played exclusively on one particular sound-system (at least to begin with), the producer thereby retaining strict control of the *production environment*, *reproduction stage* and *listening situation* variables.

(Please listen to Example CD track 5, David Bowie: *Andy Warhol*)

A fine example of *production environment* uses of pre-song mutterings which can be analysed both by perception and measurement is the dialogue between David Bowie attempting to teach recording engineer Ken Scott how to pronounce the name of Andy Warhol at the beginning of the song of the same name on *Hunky Dory* (David Bowie, 1971). In this case the producer could easily have removed this noisy addition/mistake but it was left in presumably again to add a sense of fidelity, a sense of intimacy with the recording process, and as Morey suggests, to open up the fourth wall and remind the listener that there is a production team involved as well as the performers (Morey, 2009). It is probably the case that they just thought it was funny, but nevertheless it provides interesting material for analysis regarding the social and technical aspects of production. We hear the artist seemingly unmediated by the formal presentation of the song, but what I want to focus on here are the artefacts that reveal some of the production techniques and the mechanics of what is happening in the studio during the

⁸Perry made this spoken introduction into a deliberate technique fairly early on. It features in most tracks on *14 Dub Blackboard Jungle* (The Upsetters, 2004).

recording process. How can this vignette bring us closer to this process?

Towards the end of the conversation we hear Bowie's voice accompanied by an echo, and we can guess that Bowie is hearing the same thing since he starts experimenting with the echo before bursting out laughing presumably at the strangeness of the sound.⁹ This echo is characteristic of the gap between the record/sync head and playback head on a multitrack tape recorder and Bowie is hearing this echo (delay) because the recorded sound is also being routed via the playback head to the studio monitors. This is the same technique that lies behind many sorts of delay techniques and will be described in detail in Chapter 3.3.5 and which I use in *Electronic Skank* (see Chapter 6.1.5).

In addition to this window on the process of recording to tape with an engineer speaking from a separate control room, this audible effect gives us a measurable time delay that we can use to infer the tape speed used for the multitrack recording. Trident Studios, where *Hunky Dory* was recorded, had a 3M 16-track machine and later moved to Studer A80 16-track recorders. Identifying which machine was used allows us to measure the gap between the record and playback heads. Divide this figure by the delay time and we get the tape speed which we would expect to be usually either 15 ips (inches per second) or 30 ips, and occasionally 7.5 ips.¹⁰ Measurements which do not yield exact results of 7.5, 15 or 30 might reveal the use of varispeed, or might of course lead to the conclusion that a different machine was used. It should be possible to compile a table of head spacings for all large multitrack recorders so that an accurately measured delay time might even be used to identify which tape machine was used in the *production environment*.

Once we know the tape speed we can start to question the production decision relating to the choice of this speed. A reel of two-inch tape currently costs about £200, and the cost has always been relatively high. Running tape at 30 ips gives you a recording time of about 16 minutes and of course, doubles the cost of materials compared to recording at 15 ips. Until the late 1970s when high biased tape was formulated, a big argument for using 30 ips was the better signal to noise ratio it offered, but the trade-off was a less extended low frequency response (see Figure 2.3) and a shorter maximum recording time.

If the clarity of subtle high frequency material was important a 30 ips speed might be preferred, but a 15 ips choice might imply that the low frequency instruments, bass guitar, kick drum, tuba, etc. might have been considered to have been more critical to the overall sound of the piece during the production stage.

⁹It is difficult to speak normally when you can hear your own voice delayed as sometimes happens on a bad phone-line.

¹⁰The fast delay time here allows us to assume a 30 ips speed.

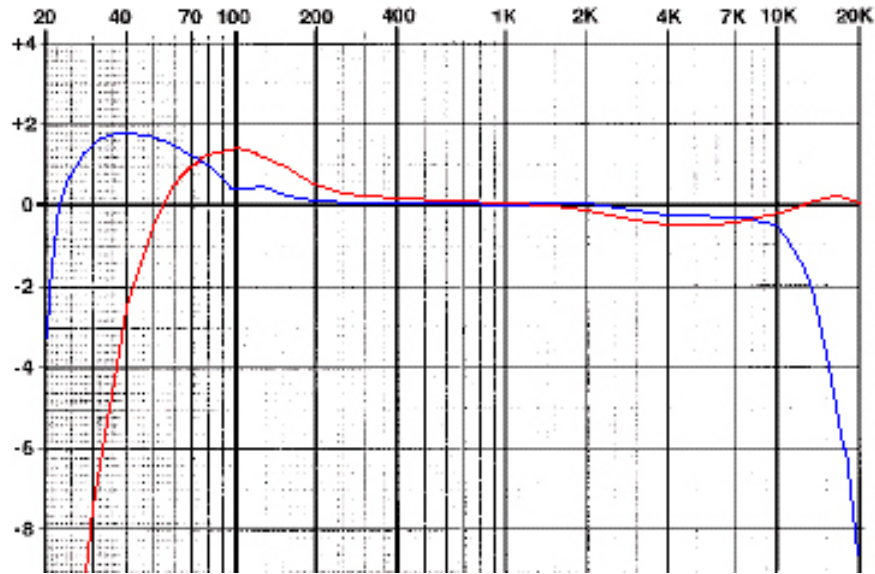


Figure 2.3: Studer A80VU Mk II frequency response curves. red: 30 ips, blue: 15 ips (Endino, 2011)

Whilst an analysis of tape speed alone is unlikely to yield unequivocal explanations of practice, it is a measurable phenomenon which may be used in conjunction with other research to help form a more accurate picture of the practice of making the music. This is a detail of the creative music practice of the recording studio betrayed by noise, and it is revealed by the use of precise technical measurements (the head spacing on a tape machine) and an understanding of how the archaeological evidence can be combined.

Glenn Gould’s humming and his creaking chair are perhaps the best known noisy traces of the human performer, and this noise is a significant factor when it comes to listening to his recordings, especially in light of his ideas about using the studio to create the perfect performance (Gould and Page, 1987). It indelibly welds performer with performed, and the result is more than a technical execution of the composer’s instructions. Gould is a central sonic element in these recordings and one must evolve a particular listening practice in order to listen to them, but once the humming is accommodated in this listening practice it would be strange to have it removed. It is clear evidence of the performer and in Waters’ sense it is an integral part of this particular “performance ecosystem” (Waters, 2007).

However, there is a process that has been developed by Zenph Studios which can emulate the particular performances of Gould’s recordings on a real piano but without the humming. Their tag-line is “Hear Glenn Gould’s playing with a stunning level of realism not heard in previous recordings” (Zenph, 2010).

Zenph must have a quite specific idea of realism and it certainly seems divorced from authenticity. This approach exhibits an exclusive rather than an inclusive sense of fidelity, yet both approaches can claim equally to be high-fidelity. Employing a more synergetic approach, the listener would accept that the “subsystem” of Gould’s humming might be contributing to the performance. In this sense the incidental noise is not incidental at all. The Zenph argument, on the other hand, would be that their process achieves greater fidelity to the *idea* of the Gould performance by taking away the disruptive noise (which does violence to the signal (Attali, 1985)), an approach that strongly aligns itself with the 19th Century composer-genius and performer-virtuoso way of thinking, and ultimately with the Platonic ideal of the musical work (Goehr, 1994).

The Zenph model of fidelity seems to lead back beyond the performance by removing selected real-world elements. It is as if the idea of the Gould performance is being captured and sonified directly, cleanly, and without being messed up by the biological characteristics of the performer - an entirely non-synergetic proposal. I think we have an example here of nostalgia, not for a phenomenon but for an ideal, and the latest technology is used to achieve this to glowing reviews from publications such as *The Audiophile Voice*, *Stereophile*, and *The Absolute Sound* (Zenph, 2010). Unsurprisingly, the praise for this approach emanates from “hi-fi” publications supporting the low-noise equals high fidelity argument in accordance with the commercial market, reinforcing this alignment between technology and practice.

2.5 Summary

Rather than making a clear assertion about the quality or value or even the definition of noise and the pursuit of fidelity, I have tried to challenge some commonly held assumptions by showing examples of very different approaches to both ideas with respect to listening and performance practice, and studio production and composition. The danger in acknowledging opposing definitions of fidelity and of noise and allowing contradictory values to be placed on them, is that both concepts become meaningless. However, although I have strong personal preferences, it is not my intention to dictate the superiority of one approach over another, and I find that even my own preferences are contingent on the composition, performance or listening practice with which I am engaged at the time. If the listener or composer is made aware of the potential for communication, both of the measurable and perceivable, in noise and imperfection, as I have begun to set out in this chapter, then they may be influenced accordingly.

Above all I have tried to demonstrate that low-noise does not automatically equate to

high-fidelity, and the interpreter who fails to interrogate and challenge the commercial pressure to adopt lower noise technologies, who is happy to be a consumer of the latest music technology, runs the risk of losing touch with what lies at the core of the music that is being interpreted. Blind (or deaf) acceptance of this link is partly what drives the adoption of new electronic and digital instruments and the abandoning of the old. Once we accept that some noisy characteristics may have positive creative benefits, we can reassess these decisions, begin to understand what has already been lost, and start to form strategies to preserve what is of potential benefit, despite or because of its noisy qualities.

In outlining some techniques for analysing noise on these three archaeological levels, I am providing a context for some of the analysis techniques used in chapters 3 and 4 and some of the practical techniques documented in chapters 5 and 6. Indeed, it is the noisy elements that have allowed me to understand some of the key features of the otherwise undocumented performance practice and technological design in the case studies. As stated in the introduction to this chapter, my focus on the *measured* is not intended to exclude the *perceived*, rather to provide evidence that may contribute to a more inclusive synergetic analysis or approach to the subject matter.

It may often be the case that a practical analysis that yields results such as the original multitrack tape machine model and speed may contribute nothing of significance to any interpretation or musicological analysis, but my interest in finding out measurable data derives from the possibility that such details may sometimes be of critical significance. Given that the passage of time exerts a marked effect on how recordings or interpretations are perceived, judged and criticised, there is every chance that some of the more obscure technical, measurable details may become highly significant at a future date. If such details remain undocumented then it is difficult to make value judgements about them, and without a solid technical foundation, some more substantial theories may remain vulnerable.

Chapter 3

Tubby's Dub Style: The Live Art Of Record Production

... [I]n a workshop where the master's individuality and distinctiveness dominates, tacit knowledge is also likely to dominate. Once the master dies, all the clues, moves, and insights he or she has gathered into the totality of the work cannot be reconstructed; there's no way to ask him or her to make the tacit explicit.

(Sennett, 2008, p.78)

3.1 Methodology

I use several techniques in order to approach an understanding of King Tubby's musical practice including interviews, analysis of video footage, and transcription of audio recordings, but the primary focus of my research has been a material analysis of the technology he used to produce records from 1972 to 1981 at 18 Dromilly Avenue, Kingston Jamaica. Since his studio no longer exists in the form it took in the 1970s some of the technical details remain speculative. However, the MCI mixing desk, the centre-piece of his studio, currently resides in the collection of the Experience Music Project, Seattle, and I have been able to examine it in its current condition. In the absence of written records or film footage of King Tubby's own studio practice, I have used footage of one of his apprentices, Lloyd "Prince Jammy" James, to work out some details of signal routing and performance practice within the Dromilly Avenue studio, but given the highly reflexive nature of this studio practice and the inevitable presence of feedback systems (both figurative and literal in the case of tape delay) some of the

most valuable insights have come from recreations of the studio setup that I have made and incorporated into my own creative music practice, particularly *Electronic Skank* (see Chapter 6.1.5).

Whilst the emphasis of this chapter is on technical detail, it should be stressed that the technology alone can do nothing, and that it is the synergistic relationship between all of the different elements: tools, techniques, performers, social conditions, and especially the “culture technology alignment” (Hill, 1988) that is responsible for making the music sound the way it does. The feedback paths between use of particular tools and techniques, the personnel in the studio during the mix, the testing of the dub-plates on the sound-system at the dance, the queue of producers outside the studio waiting to have their recordings mixed, the quality of the raw vinyl, all exert influence on the way the music on the record sounds. Far from taking a technologically deterministic position, I believe that a close examination of the tools and techniques from a material perspective can provide useful information about how affordances were exploited, how limitations were overcome, how equipment interacted, and essentially how an *alignment* of Tubby’s knowledge and expertise with the technology at his disposal shaped his own music and the musical output of his apprentices. This reflexive practice cannot be abstracted from the social and cultural environment within which it operated, and this chapter therefore presents an analysis from the technological perspective in acknowledgment that this is only one part of a much more complicated story.

3.2 King Tubby and Dub Music

Osbourne Ruddock (1941-1989), otherwise known as King Tubby, is widely credited as being one of the most influential figures in the development of the style of music originating in Kingston Jamaica in the early 1970s known as Dub. Common recording practice in Kingston in the late 1960s and early 1970s was to record drums, bass, rhythm (guitar and organ), and horns onto respective tracks of a four-track tape (the backing track), and then to record the vocals at a separate “voicing” session, often in another studio. This allowed producers to use the same backing track or “rhythm” for several different singers and even to use different lyrics, and to mix the backing tracks accordingly. Another common format was to record drums, bass, rhythm (with horns on the same track), and vocals respectively onto a four-track tape ready for mixing directly onto a mono or stereo master without the need for subsequent “voicing.”

King Tubby’s small studio in the bedroom of the house at 18 Dromilly Avenue in the Waterhouse district of Kingston was equipped only to mix these four-track tapes and occasionally to record vocalists (in the old bathroom) on top of these existing rhythms

in voicing sessions. Except for one isolated report of Lee “Scratch” Perry attempting to record drums in the tiny bathroom vocal-booth with a bass player sitting in the control room (Veal, 2007, p. 148), it was not equipped for full recording sessions. Whilst it is commonly accepted that the distinctive sound of records produced at studios such as Sun Studios (Elvis Presley), Abbey Road Studio 2 (The Beatles, Pink Floyd), or Goldstar (Phil Spector’s Wall of Sound) is the result of the specific combination of the acoustic recording space, the technology, the engineering skills, and quite often, the regular session musicians used, with Tubby’s dub mixes we can eliminate the acoustic element and the musicians’ influence as being anything more than generalized since he was mixing tapes which had already been recorded at many different studios by many different players. The two remaining constant elements therefore, that contribute to the distinctive sonic characteristics of mixes made at Tubby’s studio are the equipment used and the performance practice associated with the act of mixing.

Except for the occasional aforementioned voicing sessions, the only performers recorded at Tubby’s studio were the engineers performing the mixes: King Tubby and his apprentices, principally Philip Smart, Prince Jammy and Scientist. Given the amount of transformation and reinterpretation that happens in Tubby’s mixes and the dissimilarity between the original song and Tubby’s substantially altered mix, the mixing desk and the associated effects devices and machinery can be thought of as Tubby’s musical instruments. Defining these instruments and working out how the studio as a whole was played, i.e. the performance practice, can provide a useful perspective for a critical analysis of King Tubby’s dub style and the relationship between the practice and the design and repurposing of the instruments.

Tubby was originally an electrical engineer and crucially, in the mid 1960s, built and ran the celebrated *King Tubby’s Home Town Hi-Fi* sound system. Sound systems featuring DJs playing records as opposed to performances from live bands have been central to Jamaican dance music culture since the 1950s (Bradley, 2000), providing a nucleus for the outdoor dances and often being run as extensions to record businesses such as those operated by Duke Reid and Clement “Coxsone” Dodd. Tubby was an early adopter of transistor technology, using transistor amplifiers for the treble speakers, and valve amplifiers for the bass. He used steel horns for the treble speakers, suspending them from trees where possible, so as to project the high frequencies evenly across the dancefloor (Bradley, 2000, p. 314).

This level of care and attention to sound quality won him many clients ordering amplifiers for their own sound systems, and Tubby’s later incorporation of tape delay (and probably reverb) in his sound system controls for live spatial effects, increased the flow of orders. Very large transformers and suitable crossover networks needed for the sound systems were difficult to find in Jamaica at that time. This is no surprise

considering that Tubby and others were pushing the boundaries of speaker and amplifier technology by using the full audio spectrum from subsonic bass to ultra high-frequency treble. This meant that he had to wind his own transformers (and presumably inductors too), a task which may also have been a regular duty of his various apprentices.

Although *King Tubby's Home Town Hi-Fi* wasn't the largest sound system in Jamaica it was widely acknowledged as being the best sounding. (Bradley, 2000, pp. 314-21) His intimate understanding of frequency ranges, filtering, and speaker response needed for designing and constructing sound systems for himself and his customers, coupled with his attention to detail contributed to his extensive skills at cutting records as a mastering engineer. Bradley relates how other sound system operators such as Duke Reid and Coxsone Dodd would sometimes audition their dub-plates on Tubby's sound system rather than their own in order to hear them properly. The mastering engineer's fastidiousness is indicative of the care taken in his practice both as an electronics engineer and as a creative "Dub Organizer."¹

(Please listen to Example CD track 6, King Tubby: *Dub Organiser*)

Recording the rhythms separately to the vocals allowed producers to release different versions of the same song, but Tubby was able to take this same rhythm and with or without using the vocal track, was able to make a mix that was so substantially different to the original that it could be released as the B-side of the single, thereby removing the necessity for the producer to pay for recording a different track for that purpose. Not just an economic way to fill a B-side; if this dub mix was good enough it would become more popular than the A-side, and this soon led to the release of entire dub albums such as Lee Perry's *14 Dub Blackboard Jungle* (The Upsetters, 2004) and Augustus Pablo's *King Tubbys Meets Rockers Uptown* (Augustus Pablo, 1976), both of which heavily featured King Tubby's mixing skills.

Once he started remixing - making dub versions - he was able to start making much more creative, performative decisions in the studio, and, once cut to a dub-plate, these versions could be tested on an audience using his own sound system within a matter of hours, allowing almost instant feedback and fine tuning of his mixes. Running the sound system (controlling the *listening situation*) was therefore a key factor in the evolution of his mixing style.

Dub remixes play with the expectations of the listener as is evident in descriptions of early Tubby dubs played on the *Home Town Hi-fi*. Fidelity is a key issue in this relationship which is keenly exploited with the playfulness of the dub organiser's mixing skills and decisions as well as in the name of the sound system. A dub mix might start

¹ "Dub Organizer" is the title of a record produced by Lee "Scratch" Perry, voiced (by Dillinger) and mixed at Tubby's studio containing the line: "Tubby's are the dub organizer."

off in the same way as the original mix but at a critical point the vocals may be cut, and then reintroduced on their own with large amounts of echo, simultaneously making the audience realise that they are not listening to a faithful copy of the original song, but an original dub instead. Reports of early instances of Tubby's dub plates being played at dances invoke this idea of fidelity transgressed and the new version being demanded again and again, as if to allow comprehension and affirmation of the new original version (Bradley, 2000). The term "original" crops up again and again in vocals, toasts, titles and commentary on dub at this period, and there is a value associated with the transformation of the original, via a dub mix, into a new original. Fidelity in this context is not confined to a fixed event (record) in the past, but is allowed to be dynamic. In a dub version, the original version is often proclaimed during the track and each record almost takes the place of a live performance.

Tubby's studio remained in high demand through the 1970s with Tubby eventually taking a back seat to focus more on building sound systems, leaving the bulk of the mixing work to his apprentices. The popularity of dub reached a low point in the mid 1980s and eventually the MCI mixing desk was replaced and, after being in the possession of Rodwell "Blackbeard" Sinclair for some years, was purchased in January 2001 by the Experience Music Project in Seattle. King Tubby was murdered outside his home in February 1989.

3.3 Tubby's Electronic Musical Instruments

3.3.1 The MCI Mixing Desk

There are an astonishing number of stories and myths surrounding Tubby's equipment and unpicking these has been difficult. Putting together information from accounts by Bradley and Veal, various interviews with assistants, sleeve notes and other less formal sources, it appears that he may indeed have been using a homemade mixer, with no real multi-track capabilities, until the 1972 purchase, facilitated by Bunny Lee, of the old MCI mixing desk and Scully and Ampex 4-track tape machines from Byron Lee's Dynamic Sounds studios, formerly West Indies Recording Label (WIRL).

The evidence points to this mixing desk being designed and built by Grover C. "Jeep" Harned of Music Centre Incorporated (MCI) in the mid-to-late 1960s and it is believed to have been commissioned by Byron Lee. At this time mixing desks were not available as off-the-shelf items and they were either produced in very small runs or were more often custom made and tailored to the requirements of individual recording studios. Makers would use a combination of self-designed and borrowed circuitry and would



Figure 3.1: King Tubby's MCI mixing desk.
(Image courtesy of the Experience Music Project, Seattle)

use stock items for the more mechanical elements such as VU meters and faders, even equalisers (EQs) and pre-amps too (Ohlsson, 2004). MCI started making one of the first widely available production models - the JH400 series - in 1973, but before this, each desk would have been almost unique. Because all the signals were routed through it and it occupied the central focus of the studio, this MCI desk is the most important tool in the studio and arguably had the biggest impact on Tubby's new sound and dub style.

The desk, commonly and misleadingly referred to as a four-track mixer, has twelve input channels, each with gain, basic EQ, one auxiliary send, a channel fader and routing switches. In the master section to the right there are four output buses, each controlled by a Painton quadrant fader², there is a test-tone oscillator, monitor controls, a very

²Found on many BBC, EMI and other mixing desks from the mid to late 20th Century, these faders describe part of a circle rather than being linear tracking and work by means of stud contacts switching discreet resistors into the signal path rather than by the continuous conductive plastic or carbon tracks

early example of remote tape transport control (the coloured row of round buttons at the bottom right), and a patchable high-pass filter. The patchbay built into the side of the desk allows access to the signal path at various points in the signal chain for each channel and bus as well as access to the high-pass filter. The main sound-transforming tools used by King Tubby were the high-pass filter, volume controls, reverberation and delay.

3.3.2 King Tubby's "Big Knob" Filter

Tubby's so-called "Big Knob" filter is built into the mixing desk and controlled by a rotary switch. It is a high-pass filter³ with ten frequency steps from 70 Hz to 7.5 KHz plus an "off" position⁴ all accessible within 165° of rotation, allowing extreme sweeps to be performed with ease. It is situated at the top right hand corner of the master section and is operated by means of a 40 mm diameter skirted control knob. The knobs for controlling almost all the other features such as EQ settings, auxiliary sends etc. are a smaller 30 mm, hence the name "Big Knob" Filter.

High-pass filters are usually used to mitigate against proximity effect or to reduce low-frequency rumble, typically ranging from 30 Hz to 150 Hz maximum, so the rationale behind the enormous frequency range exhibited here is not immediately obvious. On examining the desk I discovered that the filter is a standard production model filter module made, like other components in the desk, by Altec, and not individually designed by Jeep Harned at all. This fits with Bob Ohlsson's comments about Harned building a small number of custom desks from Altec and Langevin parts prior to launching the MCI 400 series desks (Ohlsson, 2004) and suggests that the filter's inclusion was opportunistic; the lowest frequency settings being useful and the higher ones being an added bonus perhaps. Rather than designing and manufacturing an individual filter, wiring an off-the-shelf module into the patchbay and bolting it to the faceplate was a much simpler option.

The influence of Art Davis hovers over the shared designs of many EQs and filters of this period by Altec, Langevin, Cinema Engineering, and Electrodyne, and during the 1960s there were all sorts of arrangements, takeovers and relationships between these companies. Links can also be made between Davies and many of the designs that feature in Tremaine's *Audio Cyclopedia* (Tremaine, 1959), either as a good example of the trend for manufacturers using designs from component datasheets or of the use of

used in later designs. Harned's famous desk made for Criteria Studios in Miami exhibits quadrant faders on each channel, as do the EMI Redd and TG desks of Abbey Road fame.

³A high-pass filter attenuates the low frequency part of the audio signal, below the cut-off frequency, but allows all higher frequency content to pass through un-attenuated

⁴Off, 70 Hz, 100 Hz, 150 Hz, 250 Hz, 500 Hz, 1 kHz, 2 kHz, 3 kHz, 5 kHz, 7.5 kHz.



Figure 3.2: High-pass filter section. (Image courtesy of the Experience Music Project, Seattle)



Figure 3.3: High-pass filter from inside desk

widely known circuits.

This particular filter, an Altec 9069b, is a passive inductor based T-network filter, in

the same family as the Langevin EQ255a (with which it shares identical frequency step values), and the Cinema Engineering 4031. These devices comprise the class of *radiophonic* or *sound effects* filters typically used to simulate distant sound or spatial dislocation, voice mediated through radio or telephone, and other such spatial effects in radio and cinema sound design (see appendix F). The main characteristics of these filters, whether they are passive or active, is that they have stepped frequency selection and operate over a very wide frequency range. The stepped response precludes their use as dynamic performance instruments from a design point of view - indeed, one of these; the Maihak W49, positively discourages such usage (see Chapter 4), so it took a leap of imagination by people such as King Tubby (and Karlheinz Stockhausen) to use such filters in this way. This perhaps explains why nobody at Dynamic Sounds made much use of the filter while the desk was there, and extends the explanation beyond mere technological determinism. The stepped nature aligns it with a set-and-forget practice associated with traditional utility high-pass filters usually found on input channels of mixing desks. It does not encourage or suggest its use as a dynamically adjustable performance device, not least because switching frequencies results in noisy clicks which I deal with below.

Tubby's familiarity with crossovers and filters as part of his sound system work make it unsurprising that he began to experiment with this filter creatively very soon after acquiring the desk. Indeed, there are accounts of him using crossover networks with his previous homemade mixing desk (Bradley, 2000, p. 316) to split a monophonic signal (from a record or tape) into different frequency bands, allowing him to remix a mono recording by being able to attenuate the bass, mid-range and treble independently, much like using kill-switches on some contemporary DJ mixers. Bunny Lee's account of the very first experiments (Veal, 2007),(Bradley, 2000) makes it sound like an accidental discovery encouraged by Lee himself that just sounded so weird, in a good way, that it had to be repeated and immediately became part of Tubby's style, but given Tubby's experimental nature and technical experience there is a strong case that the alignment of these abilities with this equipment would have led to such experimentation eventually anyway.

Prince Jammy recounts some details about the MCI mixer:

It was a very unique board because it was custom built for Dynamic Sounds ... it had things that the modern boards nowadays don't really have, like a high-pass filter that made some squawky sounds when you change the frequency... We would put any instrument through it - drums, bass, riddim, voices. That high-pass filter is what create (sic) the unique sound at Tubby's.

(Veal, 2007, p. 114)

The clicks and crackles can be heard clearly on *Tubbys Dub Song* and in my own piece *Electronic Skank* albeit using my recreation of Tubby's filter (Portfolio CD track 5 at 6:00), also visible at various points during both videos on the DVD, particularly *Electronic Skank Edinburgh* between 2:30 and 3:00. In *Spiral* (Portfolio CD track 7 at 5:50 and 15:18) we can hear an Altec 9067b clicking as I change filter frequency settings.

(Please listen to Example CD track 7, King Tubby: *Tubbys Dub Song*)

Chris Lane refers to the filter as Tubby's "secret weapon" (Lane, 2003), but acknowledges that its use is only one of many techniques that form Tubby's style. The strongest characteristics of the filter's sound that set it apart from synthesis type filters of the same period are noise related phenomena; the discreet steps, the clicks and crunches when the frequency is switched, and the phasing effect heard when filtered signals are mixed with the unfiltered originals. Bunny Lee's description helps us to understand the signal routing in the studio:

an' Tubby's studio did ave a ting weh you could a thin it, an' do all different kinda ting with it, right, - it's not even really equalization, the ting 'ave four push-up ting, when you push the one in the middle and 'ave it up and down, with the ting, it create some mad sound, like you hear all some knife a cut thru'.

(King Tubby, 1994, sleeve notes)

The "ting weh you could a thin it" is the high-pass filter, the "four push-up ting" are the four bus-faders, and the "ting [...] in the middle" refers to either bus 2 or bus 3, and supports the idea that it was used as the filter send. To "ave it up and down, with the ting" is to adjust the filter frequency. Sending the whole mix through the high-pass filter in series would not result in audible phasing of the signal, but by sending the signal via a bus in parallel, it would be possible to mix the filtered sound with the dry sound and this would make the phase differences around the cut-off frequency audible. Each track could then be sent either in parallel or effectively only in series to the filter independently. The tape hiss (see Chapter 2.4.4) and accumulated background noise present on the original four-track tape recordings, and the spill between microphones at the recording stage all contribute to the phasing process by dramatically emphasising the phase differences between the raw and filtered signals. The tape hiss and background noise, although broad band, would have proportionally more high frequency content, and given the range of the high pass filter (up to 7.5 kHz), these phase differences would be more noticeable at higher frequencies even in the absence of programme material.

To address the issue from a different perspective, in order to create a strong phasing

sound, a useful production trick is to add some white noise (tape hiss would work very well here too) to the signal that is going through a dedicated phaser, whether hardware or software. Although a dedicated phaser works on a slightly different principle (using all-pass filters instead of high-pass filters) the addition of noise makes its effect much more pronounced. It could be argued that the addition of noise is just an attempt to make it sound retro, invoking nostalgia, since many classic phasers were inherently quite noisy, and that may be a valid reason to do so, however, it is effective either way. Because we know that tape hiss was very likely a noticeable component of the sound being fed through the filter by Tubby, and we know that there are strong phase shifts around a high-pass filter's cut-off frequency, and we also assume that signals were fed through the filter in parallel at times, we must accept that the interference between direct and filtered signals around the cutoff frequency creates a strong comb-filtering effect, i.e. phasing. This effect explains the occasional references to Tubby using a phaser.

Of course, the phaser was the secret weapon of one of the other most influential Jamaican producers, Lee Perry, specifically an early prototype of the *Mu-Tron Bi-Phase* called the *Mu-Tron Super Phasing* (Katz, 2000). This could be another source of the confusion in identifying Tubby's filter as a phaser. Perry's performance practice would need at least another chapter to analyse but a wonderfully intense superimposition of Perry's phasing and Tubby's filtering performance practices can be heard on the Tubby dub of the Perry produced track *IPA Skank* (King Tubby, 2004). In this track Perry applies strong swept phasing to the horns in the original mix which is chopped up by Tubby's high-pass filter, on this occasion configured in series with the horn track.

(Please listen to Example CD track 8, Lee Perry/King Tubby: *IPA Skank*)

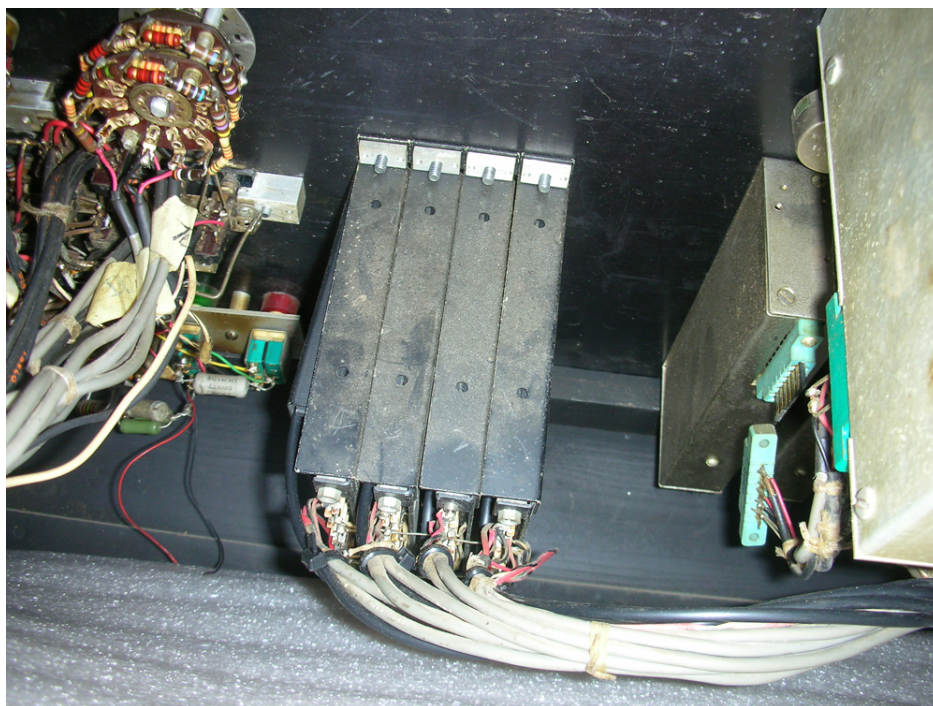
In the context of a monophonic mix the high-pass filter is also an extremely useful spatialization tool acting in a different way than delay or reverb. Being familiar with the crossover frequencies for his sound system amplifiers, Tubby would have been keenly aware of what filter settings to use to separate some sounds so that they were only projected through the suspended horns. Manipulating the filter control would then physically move the sound vertically through the dancefloor, the basslines at ground level, the high frequencies in the trees above, and adding reverb and delay would create an enormous range of spatio-temporal effects. This physical relocation of sound is absolutely consistent with some European electroacoustic and acousmatic approaches such as those developed at IRCAM, Bourges, and the WDR Studio für Elektronische Musik in Köln, but is used to engage with the listener on a more bodily oriented level due to the dance-based listening environment and the palpable sound pressure levels of the sound system. This extreme spatialisation is also normally achieved in mono!

3.3.3 Faders

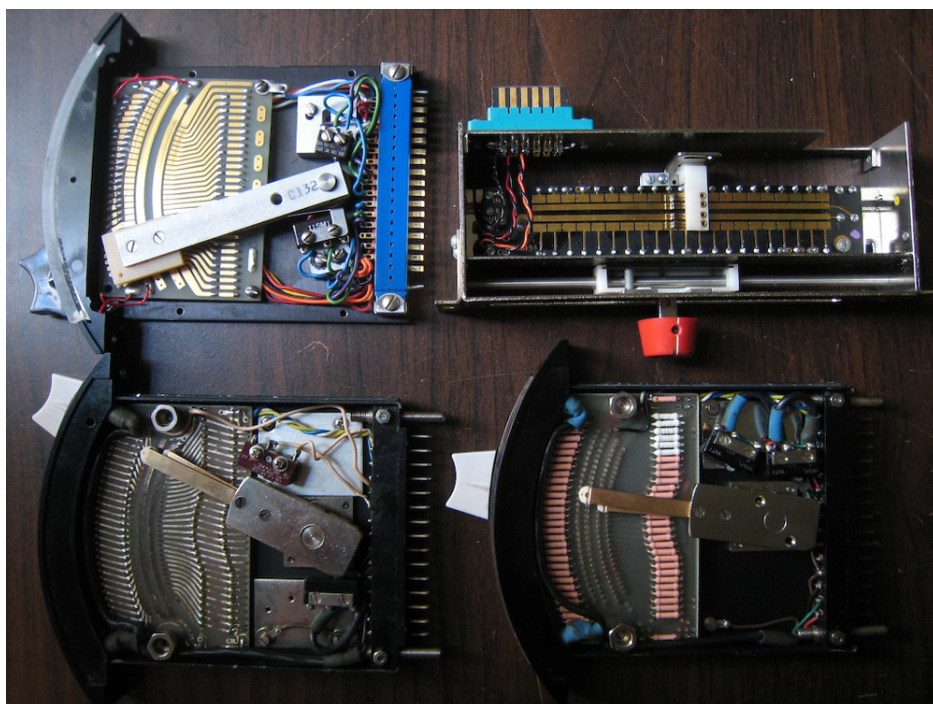
Another key performance detail is the shape of the channel fader caps. They are the round 'Rolo' style, much used by Langevin, Altec, and Electrodyne which make for a more tactile control over the volume. In the absence of mute switches the feel of the faders would have been all the more important since they were used heavily throughout a mix, and often moved very quickly with precise timing to immediately cut or reintroduce a sound. It is hard to quantify the contribution of the fader design to Tubby's mixing style, but, as the key interface between the musician and the music, this must be taken into account, as should the linear scale of the fader, marked in regularly spaced 5 dB units, thus differing from contemporary faders which tend to exhibit a more sensitive area around 0 dB as well the ability to increase gain typically by 10 dB. A qualitative analysis must be approached by acquiring some original Langevin faders and incorporating them into a performance practice but at this point I am yet to locate any with which to experiment. Since the desk is no longer functioning it is hard to tell whether the 0 dB mark on the fader corresponded to a 10 dB boost. I make comparisons between faders in more detail in Chapter 4.5.1.

Fader performance practice is rarely analysed, but a recent experience with a brand new SSL AWS mixing desk brought some details to my attention. On this SSL there is a conductive design feature that means that a fader will resist being moved unless its cap is touched with the skin of the operator. If you try to flick or nudge the fader with your fingernail, it will offer resistance and although sufficient pressure will move the fader, it will snap back to its original position once released. For my own practice, this has meant that using this desk I am unable to make delicate adjustments and nudges at particular times during a mix using techniques which I have built up over 25 years of practice, and a new technique is required. It is also impossible to use improvised techniques like resting a biro across four faders, to make a simultaneous fade on neighboring tracks. The cost of using such a desk is having to discard a prior set of highly developed skills and practice. The benefits are to me, unfathomable but bring to mind Crawford's comment: "What sort of personality does one need to have, as a twenty-first century mechanic, to tolerate the layers of electronic bullshit that get piled on top of machines?" (Crawford, 2009, p. 7).

There are many stories of Tubby replacing the faders on his mixing desk, but I found no clear evidence of any customization unless the entire top panel of the mixer has been replaced and re-engraved, which is highly unlikely. All the visible controls match the legend exactly and the only evidence that I could find in support of the idea that the faders could have been replaced was a slight variation in the shape of the heads of the two bolts used to secure each channel fader to the fascia. All the other bolts are



(a) Faders viewed from inside the desk



(b) Langevin (top right) and Painton faders. (Image courtesy of James Fei).

Figure 3.4: MCI faders

countersunk flat-headed bolts, but these are slightly round headed.

The fader caps are the red 'Rolo' shaped Langevin/Altec style, consistent with the fader modules themselves, and the legend stamped into the one-piece surface of the desk matches the scale of the fascia supplied with the Langevin faders available at the time. Even the layout of the fader module and EQ module for each channel mirrors the layout of the closely related Electrodyne channel strips of the 1960s and the whole channel layout is remarkably similar to the Electrodyne ACC-1204 console, so if Tubby did replace the faders they were either a like-for-like replacement, or he was very lucky to find a different variety that fitted exactly.

If we accept that the faders were replaced, the most likely explanation that fits with the available evidence is that he could have replaced older stepped attenuators/faders with continuous sliding faders from the same manufacturer. Such replacements would not necessarily be cosmetically noticeable since the manufacturer (Langevin) would most probably have designed such an upgrade to be a drop-in replacement. 'Rolo' caps are also very difficult to attach to other fader types. Either way, they are clearly consistent with the desk being built by Harned from parts manufactured either by Altec, Langevin, or Electrodyne, and there is no evidence at all for Tubby having replaced rotary potentiometers with linear faders on this desk. This would have entailed making a whole new face-plate and almost a rebuild of the whole mixing desk. The fader stories may well relate to modifications carried out on his previous homemade mixer. If any fader modifications were carried out on this mixing desk they were very subtle indeed, but were potentially quite critical to the ability to make deft and smooth fader movements to bring sounds in and out of the mix, especially in the absence of dedicated mute buttons. Muting could still be achieved by using the bus routing switches, but using the fader as the only control is consistent with an economy of movement exhibited elsewhere.

The quadrant faders on the buses are similar to those used on Harned's MCI desk built slightly earlier for Criteria Studios in Florida and a similar desk built for King Studios. The coloured caps, red, blue, green, and white, correspond to the coloured legend indicating the bus output connectors at the rear of the desk (see Figure 3.5), so it is unlikely that these have ever been replaced with anything but like-for-like substitutes.

3.3.4 Reverberation

The other main elements in the mix are reverberation (reverb) and delay. Reverb was routed via the single auxiliary send, accessed for each channel by a rotary control immediately above each channel fader, to a Fisher K-10 SpaceXpander (see Figures 3.6 and 3.7) - an American valve driven spring reverberation unit designed for the



(a) Bus quadrant faders (coloured tape machine remote buttons in foreground)



(b) Bus outputs, coloured legend

Figure 3.5: Bus faders colour coordination



Figure 3.6: A Fisher K-10 spring reverb unit.

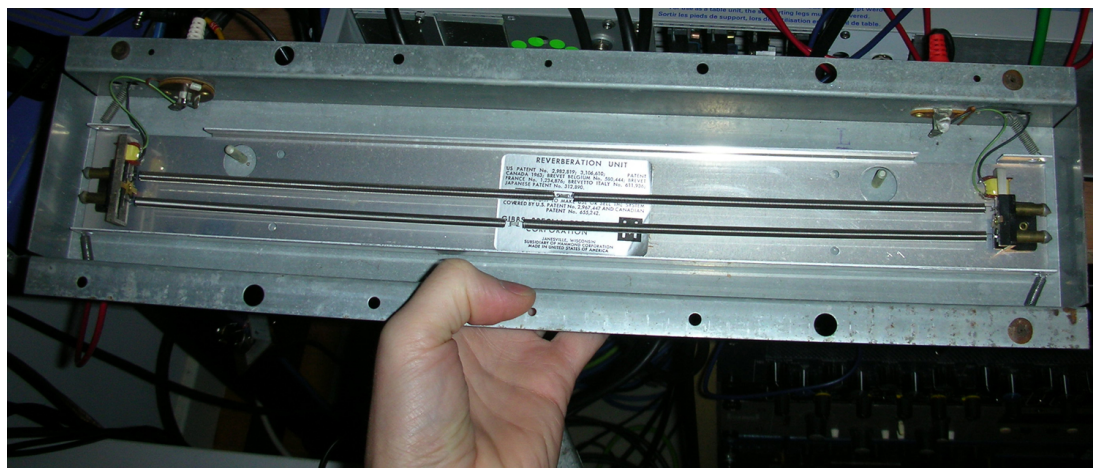


Figure 3.7: A Fisher spring reverb tank.

domestic hi-fi market to “simulate the echoes of a well-designed auditorium” (Fisher, 1967). Several accounts report that this unit was heavily modified by Tubby, but experimentation prompted by a web-forum exchange (Interruptor, 2011) has so far only revealed that muting one of the two springs in the reverb tank produces a sound closer to that heard on the records. The inputs and outputs on a Fisher K-10 are unbalanced RCA/phono sockets, and on inspecting the desk I found two cables with RCA/phono plugs hanging out of the back with the other ends hard-wired into the inside of the desk. Since the tape machines and the other outboard equipment would have been connected via professional connectors such as XLR or 1/4” jack, this increases the probability that these RCA/phono plugs might have been used for connecting the K-10. This would suggest at least some level of modification to the send source or return destination of the reverb signal within the desk, but exactly what remains a mystery.

Many of Tubby’s records feature the spring tank being dropped or knocked, and this is such a harsh sound and much louder than the normal reverb levels that it strongly suggests the use of compression, at least on the reverb return, but possibly on the main output of the mixer. Dropping or knocking the reverb tank is not a major innovation and has been used before and since in many situations. Harald Boje achieved a similar effect by shaking the amplifier for his *Electronium* (Gehlhaar and Williams, 2011). The spring tank for the reverb is mounted in the amplifier cabinet in many similar combo amps, and this is also a favourite technique of Sonic Youth’s Thurston Moore which I have witnessed on several occasions at their gigs between 1989 and 2008. The University of East Anglia had an AKG BX20 (or possibly a BX10) spring reverb unit which whilst originally being used for reverberation effects, ended up being broken by repeated use in the rough manner described above (conversation with Simon Waters, 2009). This is, therefore a use of spring reverb that is a part of a wider culture technology alignment

of abusing equipment in order to coax out different and useful sounds.

Tubby's dub of John Holt's *A Quiet Place*, entitled *A Noisy Place* featured on King Tubby's *In Fine Style* (King Tubby, 2004) is a classic example of this often used trick. The reader may observe this technique being employed in *Electronic Skank* Edinburgh on the *Portfolio* DVD at 7:54.

(Please listen to Example CD track 9, King Tubby/John Holt: *A Noisy Place*)

Since compression is a standard tool used for mastering and mixing, and notwithstanding Scientist's comments about not having used compression at Channel One studios after leaving Tubby's (Taylor, 2008), it would be highly unusual for Tubby's studio not to at least have had the option of using compression on individual tracks or the across the whole mix.

3.3.5 Delay

Before the development of digital devices in the 1980s, delay (or echo), was created using analogue tape. A signal would be recorded to tape and then played back from the same tape a moment or two later. A few dedicated tape delay machines were available from the late 1960s, but instead of using a Roland Space Echo or a Binson Echolette for example, Tubby created his delay effects the same way that many large commercial and electronic music studios did, by using a professional tape machine.

Of the two four-track tape machines Tubby purchased from Byron Lee, a Scully and an Ampex, one was used for playing back the four-track master tapes for mixing, and the other was used for delay effects. I have asked several experienced engineers and technicians including Graham Hinton (EMS, SSL Amek) and Steve Albini (Electrical Audio) which they would have used for playback and which for delay, and the universal answer has been to use the Ampex for playback and the Scully for delay because of the perceived higher quality of the Ampex machine. Chris Lane's recollections of the studio support this assumption, so I believe that delay was achieved using the Scully 4-track.

Normally, reverb and delay would be added to particular channels by using aux send controls, but with the MCI desk, in the absence of a second auxiliary send, to add delay, each channel was routed via an on/off switch to an additional output bus, which fed the tape-delay input post-channel-fader. This means that as long as the channel (drums, bass, rhythm or vocals) is heard in the mix, it is also being *sent* to the delay unit. In such a setup, the signal is recorded onto the tape with the record head and immediately played back via the playback head with the delay time being the distance between the record and playback heads divided by the tape speed. With one bus used

for the main mix and one used for the filter, this left two possible buses for delay, and in the video clip of Prince Jammy performing two dub mixes in Tubby's studio from the film *Deep Roots Music*⁵ (Johnson and Pines, 1982), you can see Jammy (who is not using the high-pass filter and therefore has three available delay send buses) using three channels for drum delay and voice and guitar delay respectively (returned on channels 1, 2 and 3). The tape outputs were returned on their own channels and the fader of each of these channels controlled the amount of delayed signal heard in the mix. These channels were also routed back, via on-off switches, to the respective tape delay output-bus as well as to the master output, thus enabling both delay level and feedback to be controlled for each delay channel by that channel fader alone. In the absence of the limitation of only one auxiliary send it is usual to have two separate controls for delay feedback and delay level.

These conclusions were arrived at through experimentation in the studio and in live performance by using a similar type of tape machine - a Studer A80 RC - and a mixing desk configured in a similar fashion to the MCI. Having incorporated Tubby's limitation of one control for both parameters into my own practice for some live performances, I found it to be extremely effective because it freed up one hand which was then able to control other parameters (see Chapter 6.1.5). The ability to have delayed instruments on their own channels allows for creative EQing of their delay sounds and the ability to create greater depth in the mix.

Chris Lane recounted to me of his visit in 1977 that whilst he and Dave Hendley were having some tracks mixed by Prince Jammy, they asked him to make a faster delay by switching playback speed of the delay machine from slow to fast (7.5 ips to 15 ips) thereby halving the delay time. Lane relates that Jammy "wasn't best pleased about this 17 year old kid interfering with his mixing style" (Lane, 2009) but that they had not used the faster tape speed before and would try it and see if it sounded good. Lane told me how he heard the faster delay used on a few records after that visit but it doesn't appear to have been used much. It is, however, used to striking effect on *Tubby's Dub Song* from Dave Hendley's *King Tubby's In Fine Style* compilation (King Tubby, 2004).

It might have been possible to vari-speed the Scully machine, thereby achieving precise delay times or modulating delay, but I have not noticed this effect in any output from Tubby's studio.⁶ The significance of this fixed delay time is that it can influence interpretation and criticism of the rhythmic qualities arising from the use of delay in some tunes. Veal attributes the double-speed drum track in Yabby You's *Fire Fire*

⁵The reader is encouraged to watch the short clip: <http://www.youtube.com/watch?v=s-KsWcv6YnM>

⁶Varispeed only became a standard feature on tape machines during the 1970s so Tubby would have had to have modified his machines to achieve this.

Dub to a conscious decision (Veal, 2007, pp. 121-2), but given the technical limitations of just two different delay-times, perhaps this effect is achieved less by design more by serendipity with the delay time accidentally being in sync with the track tempo. Either way, it still relies on Tubby's musical sense to make the decision about whether to use it in this context or not, and being able to recognize and make creative use of such an effect is a familiar technique relied upon by improvising musicians in all genres. The affordance of the delay system coupled with Tubby's experimental nature allowed such techniques to be tried and used if deemed successful. It is clear that although limited in features, the equipment Tubby was using was generally of very high quality indeed.

3.4 Performance Practice

3.4.1 Ergonomics

He do it all live, too. He don't build it up bit by bit, him just leggo' the tape and do his thing. You watch him, it like watching a conductor or a maestro at work. And of course every time it would be different. He always want to surprise people I think he even want to surprise himself sometimes and if he mix the same tune a dozen times you will have twelve different version.

(Bradley, 2000, p. 316. Bunny Lee)

The wear patterns on the desk, coupled with the footage of Jammy at the controls (Johnson and Pines, 1982) suggest that inputs from the four-track tape were on channels 7 to 10, and delay returns were on channels 1 and 2 and possibly 3, with filter return possibly on channel 11 or 12. Such an arrangement allows for a central mixing position with the four main channels accessible by both hands, delay channels operated by the left hand, and filtering by the right, with both hands able to access the reverb sends. What is striking about the footage of Jammy is the economy of movement and the agility with which the controls are manipulated. This is something more than an engineer carrying out a technical exercise at a mixing desk - it is clearly a highly skilled musician performing with a musical instrument. The limitation of only four tape channels is a liberating constraint that allowed more focus on the effects manipulation, more careful performance on the channel volume faders, and greater flexibility for one performer to structure the mix as a whole.

Sixteen-track and later, twentyfour-track recording, which became the standard in most professional studios in the 1970s, could confront the performer with a paralysis of choice

and it is perhaps no accident that the increased number of tracks adopted in later years coincided with a change of quality in dub production, not necessarily for the better. The MCI mixing desk is only 90 cm wide which allows the engineer to reach all controls easily without moving around. Compare this to a 56-channel SSL 4056 G series desk at around 4 metres wide! The possibility of parallax error in channel selection and different parameters being too far apart to be changed simultaneously are problems of larger consoles not exhibited by the MCI desk. Also, having all instruments recorded on only four tracks of tape means that many decisions have already been taken and the options for creating new arrangement and structure from the pre-recorded material are much fewer.

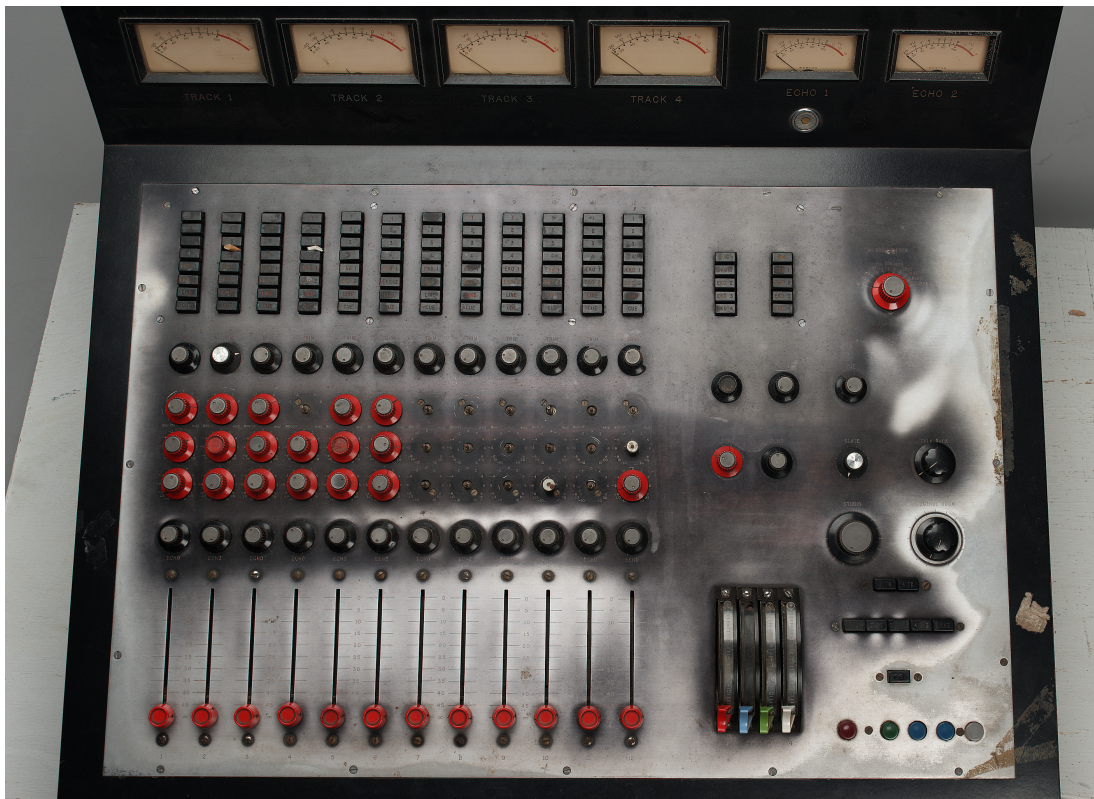


Figure 3.8: Wear patterns showing traces of performance practice. (Image courtesy of the Experience Music Project, Seattle)

For me, the most exciting physical evidence of performance practice are the aforementioned wear patterns on the surface of the desk, particularly around the filter control, with clearly visible traces of thumb, fingers and the palm of the right hand indicating heavy usage (see Figure 3.8). It brings to mind the Fender signature series of guitars such as, Andy Summers' Telecaster and Jaco Pastorius' Jazz Bass and, in the absence of any film footage of Tubby himself, provides the clearest visible, archaeological evidence of Tubby's performance practice.

3.4.2 Practice Led Research

In order to work out some of the finer details of Tubby's performance practice it was necessary to try to emulate his studio setup and test various theories. The main considerations were accurately creating the conditions for reverb, delay and filtering. I sourced a Fisher K-10 Spaceexpander from the USA but was unable to find a Scully four-track machine so for delay I used a Studer A80 two-track machine instead. Since the head spacing (which defines the delay time) is similar, this was a reasonable substitute. I built a varispeed controller for the Studer (see Appendix B.2) to fine tune the delay time and also to allow for dynamic delay time changes. The latter option was not a technique used by Tubby, but it was something I wanted to use in my own performance practice.

I designed and built the filter used for early experiments and for my piece *Electronic Skank* (see Appendix B.1) before I had located and examined the original mixing desk. My version sounded good but I have since sourced an original Altec 9067b filter (a 9068b low-pass and 9069b high-pass in one unit) and this has become part of my regular setup.

After some calibration my filter was suitably similar in sound and almost identical in physical performance control so that it could be combined with the other devices to test the theories of practice I had developed. The assumptions were duly checked and comparisons made with the Prince Jammy footage, and a reflexive process of reconfiguration, performance and observation contributed to the conclusions contained in this chapter.

It was not until August 2010 when I visited the Experience Music Project that I was able to examine the inside of the mixing desk and discover the exact filter model used - the Altec 9069b. Some further research confirmed the frequency values of the Altec as being accurately represented on the legend on the mixing desk, but it took another six months until I was able to locate an identical model to carry out tests on. It then took a further twelve months to find the courage, at the risk of destroying the filter, to open up the sealed unit in order to inspect the circuit design, although as soon as I saw the filter and the diagram on the label, I was confident of the general circuit design but not the component values.

On dismantling the Altec 9069b (see Figure 3.9) I was disappointed to find that the capacitors are housed in a sealed plastic unit (the green cylinder on the left of the image) and are therefore not directly observable. The inductors, on the other hand, are toroidal multi-tapped devices, with enough taps on each that two inductors cover the entire range of values needed for the unit. Toroidal inductors offer good suppression of

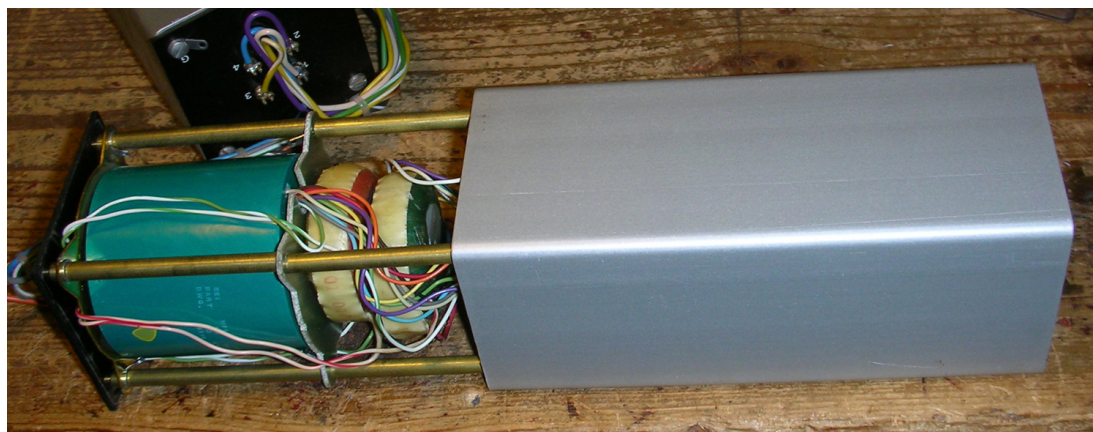


Figure 3.9: Inside the 9069b filter.

interference. The switch is a three pole make-before-break rotary switch, as predicted simultaneously switching two capacitors and one inductor tap for each stepped position. This directly accounts for the resonant clicks when the frequency is switched - a quite different quality of clicks exhibited by my active filter.

Without wanting to destroy the unit by breaking open the capacitor housing to find out component values, I observed the capacitor values of a very similar Altec 9065 fixed frequency high-pass filter and was then able to work out the inductor value based on the cutoff frequency. Having noted the large number of Cinema Engineering devices pictured and documented in Tremaine's *Audio Cyclopedia* (Tremaine, 1959) and being aware of the strong links between Cinema Engineering, Langevin and Altec, I found a table of component values for constant- k filters in Tremaine with which the component values of this 9065 fixed filter were absolutely consistent. It is reasonable to assume that the component values in the 9069b filter as used by King Tubby are also consistent with the values in this table for a constant- k filter.

A short Altec technical paper (Noble, 1966) directly addresses the issue of impedance matching and the consequences of the lack of any internal damping circuit in this design. The main effects of different input and output impedances are a change in the steepness of the filter slope, from between 12 db/oct to 18 db/oct, and a dramatic increase in filter resonance with high load impedance. What this means is that even though the filter has only one adjustable parameter - frequency - its sound may be changed dramatically according to how a signal is sent to it, and what sort of device is fed by the filter's output. An understanding of the principle of impedance helps the practitioner to arrive at the desired configuration, and in my own practice I have found that different load impedances offer useful results in different circumstances. In general I follow the output with a low impedance volume pedal thus keeping the resonance controllable but still noticeable. Without knowing the input and output impedance

characteristics of Tubby's MCI mixing desk, it is hard to predict the filter response, however, the nearest value to a standard during the 1960s and 1970s was an impedance value of $600\ \Omega$ so this figure can reasonably be used as a working value. This is the value recommended by Noble (Noble, 1966). Of course, fine tuning and experimentation by ear was necessary to achieve a similar sounding filter response, but it was not as simple as just connecting the device and considering the job done. Even in the context of a measureable, material analysis, this is a clear example of synergistic behaviour as outlined in Chapter 1.6.

My portfolio piece *Electronic Skank* is based on the tools and techniques described above and as well as being a straight piece of structured improvisation for small ensemble, as part of a reflexive process it has been a valuable means of testing the interaction of these practices in a live environment (see Chapter 6.1.5).

3.5 Summary

Tubby's feel is sometimes ascribed to his love of jazz (Veal, 2007, p.117), and the improvisatory nature of his mixes supports this theory. Bunny Lee: "if he mix the same tune a dozen times you will have twelve different version." (Bradley, 2000, p. 316) Tubby fixes his improvisations in the form of records, and he draws on a number of structural, spatial, rhythmic, and timbral techniques to stamp his identity onto each version.

In *Rebel Dance* (King Tubby, 1989a), an analysis of which is presented in Chapter 4.5.2, he uses the clicks and crunches as the filter is switched between frequency steps to punctuate and augment the rhythm, at one point imparting a triplet feel. This would not be possible with a continuously variable synthesizer filter but it is also clear that it is not simply technological determinism at work here either. It is the *alignment* (Hill, 1988) of Tubby's tacit knowledge of electronics coupled with his musicality and the affordance of the instrument characterized by the clicks and steps, which combine in his practice and which make it unique and which allows him to make significant musical changes to a tune's internal rhythm. An argument can be made that tape delay is also used to re-structure rhythm and to create cross-rhythms, but in performance terms this is perhaps less deliberately controllable and is certainly less performative since the delay time is limited to one of two values, whereas the filter can be stepped between its eleven frequency steps at will to create precisely timed rhythmic interventions.

It is sometimes difficult to distinguish between Tubby's and Jammy's mixes but this can be partly explained by remembering that Jammy was Tubby's apprentice and that repeat business for the studio revolved around a house style. Rather than the technology

solely determining this style, I suggest that social and economic factors would have encouraged the convergence of each engineer's mixing style *within* the framework of limitations and affordances set by the available technological configurations. It is beyond the scope of this chapter to analyse the different mixing styles of Tubby, Jammy, Scientist and the other apprentices, but such a study could build on the research presented here and explore the preferences and refinements of each engineer's practice within the context of a common set of tools and instruments.

On examining the instruments, and in particular the mixing desk, although most were of very high quality, it is clear that Tubby had to deal with and overcome severe limitations, and while it is certainly the case that some of the equipment lent itself to being used in a particular way, it was Tubby's expertise and creative imagination that exploited the affordances of these elements and combined them into a single musical instrument enabling the production of such inventive and enduring music. A grasp of these technical characteristics and limitations is essential for a complete musicological analysis of Tubby's creative music practice. This material research offers as much detail as possible from a technical perspective but a deeper understanding of King Tubby's music is only possible by considering socio-economic and other factors as well.

In recreating the instruments and their configuration, I have become convinced that a degree of electronic knowledge is essential for the practicing electronic musician in order to create a deliberate sound. A good understanding of the basic principles is essential in the interpretation of existing works, and is of considerable value in the composition of electronic music that is able to last and be interpreted in the future. In the case of the Altec 9069b, even using an original unit will not necessarily lead to a Tubby-like sound unless the effects of input and output impedance are understood. If the composer knows why she is using particular equipment in a particular configuration then that can be communicated. If the resulting sound is just a fortunate coincidence then the score or documentation is unlikely to capture this, and when the future interpreter is unable to source exactly the same equipment then some vital characteristics may be missed or lost entirely. Even by sourcing the same equipment, as I have demonstrated, the sonic results are not guaranteed, and that is before technique and performance practice are even considered. My research down to the component level seeks to identify the relationships between devices and instruments so that in analysis they may provide insight into the composition and performance practice, and in performance and composition they may be used for greater creative expression. Understanding the instruments and their configuration is the first step towards understanding the related performance practice. It is clear that King Tubby had that depth of understanding right down to the component level and was able to align that with a musical practice that continues to influence musicians today.

Chapter 4

Stockhausen and the Maihak W49 HörspielVerzerrer

It's extremely important to comprehend works, which were born to a particular historical moment, for their uniqueness. It just won't do to be continually discarding everything and making something different, but rather we should be preserving things and adding new ones. Anyway, it is my experience of music that every instrument, every item of equipment, every technique can produce something unique, which can be achieved in no other way. Since that is the case, then we can speak of an original technique, and thus deal with an original instrument.

(Stockhausen and Kohl, 1996, p. 97)

4.1 Methodology

I was initially drawn to making a comparative analysis of the styles of King Tubby and Karlheinz Stockhausen because I was intrigued by their similar use of the stepped filter, a device so seldom used as an instrument that I have found no other obvious examples. I suspected that such an analysis might reveal some common principles of practice that could be applied to electronic music of any genre, and I hoped that each case study would illuminate details about the other. In this chapter I first present a case study of Stockhausen's use of stepped filters in one piece of music, alluding to the previous chapter along the way. At the end of the chapter (Section 4.5.2) I highlight the congruities and differences between the practices of both Tubby and Stockhausen and offer some concluding observations.

In contrast to the investigation of King Tubby's instruments and practices, there is a much greater amount of documentation about Stockhausen's methods, so in this chapter I am able to investigate the use of a particular filter, the Maihak W49 Hörspielverzerrer, in a particular piece, *Mikrophonie I* (Stockhausen, 1974). The score, which includes photographs and frequency tables, seemingly provides all the information needed to communicate the essentials of the performance practice and instrumentation for any future performances to retain the sense of authenticity alluded to in the quote above. However, on closer inspection of this and the many other papers, interviews and essays on the subject, and listening to different performances by different ensembles, a number of problems arise that deserve our attention not least of which is a shifting attitude towards certain noisy aspects of the instrument by the composer himself.

The materials I am relying on to pursue this analysis of Stockhausen's practice are the score for *Mikrophonie I* (Stockhausen, 1974), the recordings, a live performance, technical documents, published articles and interviews, two interviews I conducted with Rolf Gehlhaar - Stockhausen's assistant between 1967 and 1970, an interview with Volker Müller - engineer at the West Deutsche Rundfunk Studio Für Elektronische Musik (WDR Studios) from 1971 to 2005, and practical work measuring, analysing and reconstructing the W49 filter itself. As with the King Tubby analysis, the question of fidelity, both in its low-noise sense and in its authenticity sense (see Chapter 2) is raised at various points, and the issue of noise is also central to the different attitudes towards the use of the filter.

My own practice has been engaged with this research, not by building a fully functional replica of Stockhausen's filter as I did with Tubby's filter, but by interrogating the crossover between Tubby's and Stockhausen's sound worlds by employing an Altec 9069b - King Tubby's filter - in my own performances of pieces by Stockhausen that might have used the Maihak W49 (see Chapter 5). Additionally, I found some Maihak W66c faders and was able to customise one unit to recreate the two-faders-in-one-slot interface found on the W49 (see Appendix G and *W49 Mockup.pdf* on the Portfolio DVD). This approximation of the control interface section only was useful in allowing Rolf Gehlhaar to demonstrate specific performance techniques (See Section 4.4.2 and *Rolf Demo.mov* on the Portfolio DVD).

4.2 *Mikrophonie I*

In 1964 Karlheinz Stockhausen wrote one of the first European pieces of music for performance combining acoustic instruments with live electronic instruments.¹ Other

¹Kagel's *Transición* (1959) is generally recognised as the first substantial such piece.

pieces had been performed using recorded electronic sound including his own *Kontakte* (Karlheinz Stockhausen, 1960) - a tape piece performed in one version with live piano and percussion - but *Mikrophonie I* was an attempt to redress the balance between fixed sounds and fluid performance. The biggest problem when combining fixed media sounds with live performance was the one-way nature of the relationship between the elements. Each musician had to slavishly follow the tempo dictated by the recorded medium as that recording, aside from mainly amplitude and panning adjustment by the sound projectionist, could not adapt or react to the performers in a reciprocal fashion. Stockhausen wanted the electronic elements to play a complete role within the performance, both active and reactive, in the same way that the acoustic elements behave, and using some of the tools familiar to him since the 1950s he conceived of a way to do just that, by liberating the formerly studio-based electronic sounds from the fixed medium and bringing them into the live performance environment.

Stockhausen used the Maihak W49 filter in the realisation of *Kontakte* between 1958 and 1959, and the composition and performance of *Mikrophonie I* from 1964 onwards. The filter is often present in photographs of performances of many of Stockhausen's pieces in the 1970s and there is evidence for its use in performances of *Kürzwellen*, *Prozession*, *Hymnen*, *Aus Den Sieben Tagen*, *Für Kommende Zeiten* and many other pieces played throughout the 60s and 70s.

He was based at the WDR Studios in Cologne for a large part of his career and was able to draw on their huge technical resources and the expertise of their engineers and technicians whilst working there.

The instruments used to create these early works were by necessity mostly laboratory test equipment, including oscillators, filters and meters, repurposed as musical instruments. In the mid 1950s there were only isolated instances of dedicated electronic musical instruments but by 1964 when Stockhausen was writing *Mikrophonie I* the use of electronic instruments was far more widespread, with dedicated electronic music studios in many major cities and universities throughout Europe and the USA (Davies, 1968). *Mikrophonie I* was another step forward in its combination of acoustic and electronic sound manipulation with live performance in the classical tradition, i.e. using a score and being presented in the concert hall. This piece evolved from Stockhausen's experimentation with a large tam-tam in his garden whilst a colleague, engineer, Jaap Speck manipulated a filter and amplifier to change the sound picked up from a microphone, recording the results onto tape. Stockhausen's recollections of the surprise in hearing these recorded experiments played back mirror the excitement related by Bunny Lee at Tubby's first experiments with the Altec 9069b filter (Kurtz, 1992, p. 135) and (Bradley, 2000).

In a performance of *Mikrophonie I* there are two groups of three performers. With the tam-tam placed sideways in the middle of the stage, one group is on the left hand side, the other on the right, a spatial division which also extends to the loudspeakers. In each group, one stimulates the tam-tam by various unorthodox methods including hitting, scraping, bowing, shouting; one uses a microphone to pick up sounds at varying distances from the surface of the tam-tam and from the site of stimulation, also using a resonator such as a cardboard tube, box, or wine glass to acoustically filter the sound; the other sits in the audience and controls a filter and two volume faders, changing the timbre as well as the amplitude and position in space of the sound picked up by the microphone. The loudspeakers are positioned at four corners of a square, with each group's filtered sound being projected through front and rear loudspeakers on the corresponding side to their stage position, giving the audience an immersive sound. The sounds heard by the audience are a combination of acoustic sounds emanating from the tam-tam itself as well as filtered sounds amplified through the loudspeakers. Rolf Gehlhaar who performed the piece with the Stockhausen Ensemble between 1967 and 1970 describes the process:

...the tamtam is so strong that the loudspeaker sound has a bit of a struggle sometimes, and if it doesn't come out of the loudspeakers then the filters are useless. The beauty of the piece is exactly that polyphony between the amplified and the filtered - that's why the filters are important because you get an amplified sound which is different from the unamplified sound, so it's quadraphonic, or polyphonic. (Gehlhaar and Williams, 2011)

This description of an immersive, multilayered sound is immediately reminiscent of the sound world created in King Tubby's dubs, and the presence of filtered sound alongside unfiltered sound also has a direct parallel. The slight delay and phase-based anomalies heard in the hall have similar properties to the sound interaction created through Tubby's use of parallel filtered signals. (see Chapter 3.3.2)

4.3 The Maihak W49 Filter

4.3.1 Score Documentation

Thanks to the extensive documentation in the scores for both *Mikrophonie I* and *Kontakte* (Stockhausen, 1968), we can get a very good idea of how Stockhausen intended this instrument to be used and what it was actually used for. The *Mikrophonie I* score gives precise directions for each performer and also offers additional information about the choice of implements with which to excite the tam-tam (Stockhausen, 1974). Great detail is given about the electronic instruments used for performing

the piece, and this was the starting point for my examination of the Maihak W49 Hörspielverzerrer.

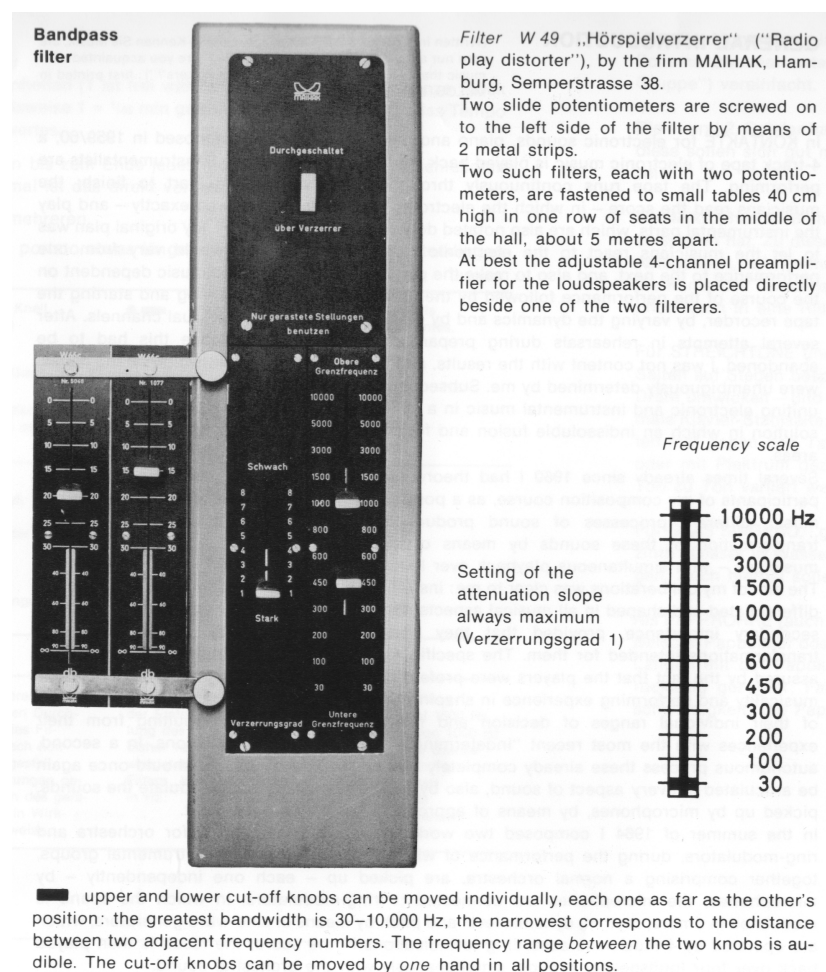


Figure 4.1: *Mikrophonie I* score extract: W49 with W66c faders attached. (Stockhausen, 1974)

In Figure 4.1 we see the W49 filter with two Maihak W66c volume faders bolted to it, together comprising the electronic performance instrument used by each of the two performers in *Mikrophonie I*. On its own the W49 weighs a hefty 10.9 Kg and accepts a balanced input signal which passes through a low-pass and a high-pass filter to a balanced output. Filter slope and frequencies are adjusted in steps by means of three sliding switches. From the filter's output the signal is split and sent to the two W66c volume faders which govern the amount of filtered signal sent to a front or rear amplifier and loudspeaker.

In the first performances and recordings as well as the video of *Mikrophonie I* one filter and potentiometer set was operated by Stockhausen himself but the other set was operated by Hugh Davies (potentiometers) and Jaap Speck (filter). With the potentiometers attached to the filter as pictured in the score, this would have been

difficult for two people to operate simultaneously, and since Gehlhaar took over from Hugh Davies as Stockhausen's assistant in 1967 after Speck stopped working for Stockhausen, this is consistent with his claim to have come up with the idea to join the units together, thus relying on their combined weight to stop the potentiometers from moving around, and thus creating a new instrument which was to be used for many years.

RG: I remember a change we made. I remember something, that when I came and we started, just before we started rehearsing we made a small change in the setup and I think ... because I remember I made the prototype but I can't remember exactly what it was. It had to do with the two potentiometers. I think before there was the filter and there was the potentiometer box, and then I had the clever idea of making some brackets and attaching the potentiometers to the filter box, so that it was one unit.

And I think, the two little brass rods that connect the two; I made those.

Because that was the first thing I said when we were rehearsing, I said you know it's impossible because this thing moves [potentiometers] the W49 is heavy and it doesn't move, but this thing, you know, so why don't we just attach it.

SW: So you designed the bolting it together?

RG: Yes. Big Deal!

(Gehlhaar and Williams, 2011)

The score describes this modification: "Two slide potentiometers are screwed on to the left side of the filter by means of 2 metal strips," (Stockhausen, 1974) but I think that Gehlhaar underestimates the significance of this act. From this point on, all photos featuring the W49 in use by Stockhausen show the two W66c faders attached as in Figure 4.1 and the decision to do this has therefore had repercussions for many years, indeed, all five W49 filters I examined at the WDR Studios showed evidence of having been customised in this way (three are seen in Figure 4.4). The choices that might have been struggled with subsequently have been avoided by the existence of this combination of tools as an integrated musical instrument, the limitations and characteristics of which must have influenced composition and performance practice during this time. This arrangement is perfectly set up for left hand control of volume and right hand control of filter frequency.

4.3.2 Technical specifications

In assessing the impact any technical details may have over the sound of the piece, we must first consider what is specified in the score and how those specifications match up to material research. Firstly, the W49 appears to have been made in very small

numbers, perhaps not more than 120 units. Of the handful of W49s I have seen (Serial Nos. 6, 16, 21, 25, 34, 69, 071 and 119) all except one (Serial No. 071) have been consistent with that pictured above and described in the technical documentation. Costing around DM8000 (Stockhausen, 1972, p. 170) in 1969,² these were only affordable by the large broadcast companies such as West Deutsche Rundfunk. There is no suggestion that the W49 was widely available as a commercial item hence the assumption by some that it was custom made for Stockhausen. Stockhausen himself states that he had the filters custom made but it is more likely that the units were custom ordered rather than custom designed; some paperwork relating to the costings for the 1970 World Expo project includes quotes from Maihak GMBH for the purchase of multiple units.³ It could also be that he is referring either to Rolf Gehlhaar's use of two brass strips to fasten the W66c faders to the W49 filter (Gehlhaar and Williams, 2011), making the instrument in Figure 4.1. The design date of the circuit diagram is 12th July 1950 which precedes any of Stockhausen's electronic music experiments (Maihak, 1951).

The W49 with serial number 071, from the collection of the Musikinstrumente & Design Online Museum in Berlin, seems to have been supplied by Maihak with an extended upper frequency range, reaching to 14 KHz instead of 10 KHz. Indentations of the original legend are just about visible alongside the over painted revised details, but every indication suggests that this was not an after-market modification since the entire panel legend has been overprinted, not just the frequency numbers.

The original Braunbuch datasheet (Maihak, 1951)⁴ describes the W49 as a "Verzerrer für Hörspielzwecke," i.e. a distorter for radio play purposes. According to this document it was designed "for the creation of acoustic effects by electronic means using frequency cutting." [my translation]. Some time after taking over Maihak's audio division, Telefunken also produced a short undated document containing some of the specifications of the W49. This is in German, French and English, and notably includes some key phrases in the French and German not translated into English. The most obvious is as follows:

German version:

Der Hörspielverzerrer W 49 ist ein regelbares Dämpfungsglied, mit dem der Frequenzumfang von Sprache und Musik verändert werden kann.

English version:

²Compare this with \$125 in 1963 for the Langevin EQ 255A, almost identical to the Altec 9069B (Souther, 1963)

³Hand typed correspondence from the 1970 Osaka file in the Stockhausen Archives, Kürten.

⁴This document took me four years to track down and is included in the digital resources attached.

The Attenuating Equaliser W 49 is a four-terminal network with variable frequency response.

(Telefunken, Undated)

Here, the German version suggests usage of the W49 as “an adjustable attenuator with which the frequency range of speech and music can be modified” [my translation].

The German name “Hörspielverzerrer” translates literally as radio play distorter. The French name “Correcteur de son pour pièce radiophonique” translates as sound correcter (equaliser) for radiophonic productions. In the technical document, the English name is simply “Attenuating Equaliser,” leaving no implicit cue as to its intended usage, however, it is clear that it was designed for transforming sound in the context of radiophonic sound design, such as simulating distance, telephone conversations etc.. This filter is one of the best examples of the European type of radiophonic filters that includes devices such as the Eckmiller HV-53, HV-55 and W86a. (See appendix F).

This usage is still rather open-ended, but printed clearly above the frequency switches and unmissable to any user is the following instruction: “Nür gerastete Stellungen benutzen” i.e - only used detented settings.⁵ This is where Stockhausen’s will clashes with the original purpose of the design since his score for *Mikrophonie I* instructs the filter performers to continually adjust the filter frequencies in direct contravention of the legend printed on the instruments in front of them. The consequences of this transgression will be revealed below, but we should also consider this repurposing in the context of the other performers in the piece. *Mikrophonie I* also involves two performers scraping, bowing, shouting into, and doing all sorts of other things to a tam-tam that diverge from the expected way of using such an instrument, and two other performers using microphones in a similarly unorthodox manner. The repurposing of the W49 filter is therefore consistent with the approach demonstrated throughout the piece in which Stockhausen is pushing each performer to extend the music-making potential of each instrument or tool. We could describe this as an alignment of the avant garde tendency to explore new sound worlds with the repurposing practice to explore new technique.

4.3.3 Transmission Noise

In the case of King Tubby’s filter we faced the problem of a lack of written information and the inevitable discrepancies between different accounts of the technology and practice from different people. Surprisingly (to me, at least) we encounter similar

⁵The sliders click into position at each frequency step.

discrepancies in written and technical accounts of the W49 and its use, sometimes even from the same source, and often due to the passage of time perhaps colouring memory. Although not strictly one of the core types of noise which we are using for analysis, any corruption or confusion of the information contained in a transmission can be considered as noise for the purposes of examining a few details in the documentation relating to the W49 filter. As with the Tubby case study, unpicking these details can be critical to arrive at an accurate understanding of how the instruments functioned and sounded.

A telling detail is the omission, in the Telefunken technical document, of one of the frequency bands from the upper limit list (Telefunken, Undated). When taken in conjunction with the “circa 25 dB/Oct” filter slope indicated in the *Mikrophonie I* score (Stockhausen, 1974), this calls into question the technical document’s quoted figures of 40 dB/Oct attenuation. In fact, derived from standard one pole RC filter networks, filter attenuation is usually expressed as a multiple of 6 dB/Oct, so the score figure of 25 dB/Oct could be explained as a metrically minded rounding error of the standard 24 dB/Oct, but that leaves a problem explaining the 40 dB/Oct figure. An alternative way of quoting filter slope characteristics is in dB/Decade, and since 40 dB/Decade is equivalent to a 12 dB/Oct slope (nicely divisible by 6), it seems that the Telefunken document might be in error in this instance. The original Maihak document states an *overall* signal attenuation of 40 dB, and the filter slope diagrams show a filter slope of 12 dB/Oct.⁶ One can see how two 12 dB/Oct filters in series might be assumed to add up to one 24 dB/Oct (40 dB/Decade) filter, giving rise to the mistakes.

Another related inconsistency occurs in Stockhausen’s lecture on electronic music given at the Institute for Contemporary Arts in London in 1971 (Stockhausen, 2007). He describes the filter frequencies ranging from 30 Hz to 12 KHz in nine steps whereas the legend on the early version of the filter describes the frequency ranging from 30 Hz to 10 KHz, and all known versions have eleven steps, not nine. It could be a simple inaccuracy but it could also be evidence of either a noticeable difference exhibited between the specified frequency response and the observed frequency response of his particular filters. Although there is nothing else that I have found to support the hypothesis, it might even be evidence of customisation of the filters carried out at the WDR studios. For the purposes of the lecture and describing the three levels; high, middle, and low, citing nine frequency steps (i.e. easily divisible by three) is either a prudent move to aid clarification or just a simple error, the main point being to emphasize the stepped nature of the filter. This indicates that the precise number of filter steps was clearly of less importance than the range of these steps and their division into three rough groups.

⁶The full Maihak datasheet is included on the Portfolio DVD.

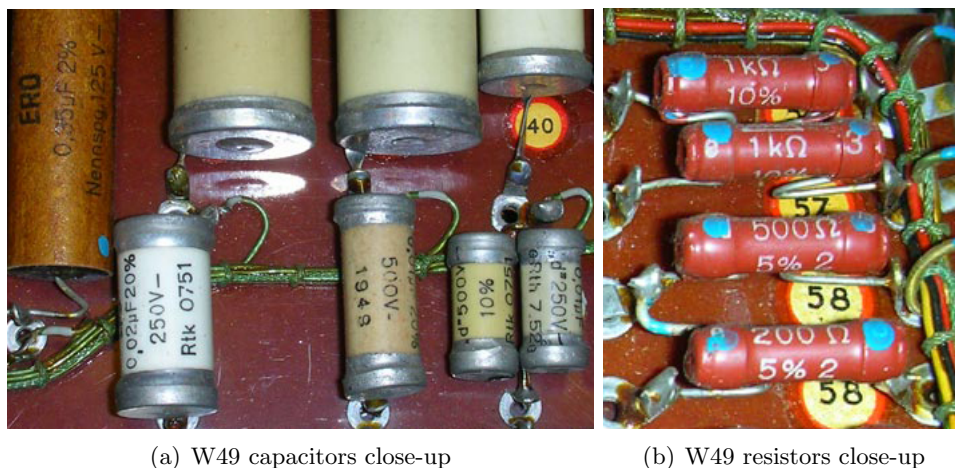


Figure 4.2: Maihak W49 component tolerances

Once we open up the filter and peer inside we can see that the many capacitors and resistors have tolerance ratings of between 2% and 20%, although most are either 2% or 5% (see Figure 4.2). This introduces the possibility, in the absence of component selection or matching, for each W49 to exhibit different sonic characteristics from the factory with regard to frequency settings, departing from the technical specifications again. Allowing for changing capacitance over time, each instrument will take on its own individual sonic characteristics, and will, as is the case with serial number 071, eventually fail. Between manufacture and failure the instrument will exhibit a range of different sonic characteristics and this variability must be allowed for in any frequency response measurements made on surviving units. I have been unable to get an opportunity to take any such measurements due to the scarcity of these instruments.

4.3.4 Incidental Noise

Perhaps the most important and easily overlooked characteristic of this device is that the faders are not actually continuous faders in the normal sense - they are linear switches. According to the legend printed above the faders the unit was not designed for people to move the faders around dynamically, so the resulting clicks and scrapes when doing so are not the signs of a poorly designed instrument, but are inescapable factors of the design of this particular device, and, critically, are factors of the repurposing of the instrument for electronic music performance - the realignment of the instrument with a music performance practice.

In the score Stockhausen demands: “2 bandpass filters with variable bandwidth (stepped, click-free)” (Stockhausen, 1974). Much of the filter notation, however, is

continuous rather than stepped, but he also states: “... all filter changes notated as graphically continuous actually occur stepwise, due to the stepped filter” (Stockhausen, 1974). Continuous lines therefore suggest dynamic sweeps and stepped notation suggests the use of the filter steps as rhythmic elements, especially when the notated step coincides with a tam-tam event. We can also observe a similar technique being used to control the oscillator frequency in *Mantra*, with clearly notated fixed frequency steps as well as long and improvised glissandi (Stockhausen, 1975), a key difference being that the oscillator/ring modulator used for *Mantra*, the Modul 69b designed by Peter Lawo, was designed specifically with this use in mind.

He goes on to describe an “EXAMPLE of the division in Hz of the “W49” filter used so far” (Stockhausen, 1974), tentatively leaving open the possibility of substituting the W49 for a different filter in the future, and allowing other performers the scope to interpret the score by using different devices. However, since the stepped filter has not been adopted as an instrument by enough people to make commercial production a viable proposition, there are no real alternatives, and W49s are so rare as to be practically impossible to procure. Between the 1965 composition and the 1974 publication of the score there is still evidently some attachment to the old W49, possibly because there were no alternatives, but it is characteristic of Stockhausen to demand better (click-free) technology, even though this attitude softens over time as we shall see below.

What comes across from the performers of *Mikrophonie I* is an initial frustration with the audible clicks and the physical resistance of the faders. Gehlhaar recounts how they had to use contact cleaner to get rid of the frustrating clicks, and also how their fingers were painful and tired after a performance. He did not speak about the filter fondly (Gehlhaar and Williams, 2008). Stockhausen himself continued to use the filter for many years so perhaps he either did not share Gehlhaar’s reservations or, through extended usage, he developed a practice which overcame these physical restrictions. The departure of the performance from the “click-free” demands of the score is subject to Stockhausen’s post-rationalisation in a lecture delivered in 1991:

They were so-called Hörspiel-Verzerrer W49, built in-house at the WDR in Cologne: filters with carbon strips. It is really interesting how very old-fashioned that sounds (after all, violins with catgut are used today). Such materials are glorious, aren’t they? The two metal levers of the filters scrape along on the carbon strips, and spray must now and then be used... Today, if you try to substitute computerized filter simulations, the characteristic sound goes to hell. The scraping and the skips between filter-levels is lost; but they actually belong to such a sound, when it is brightened up from below to above, or vice-versa. The score is also written in such a way that both controls can be opened and closed in the span between the index-finger

and the little-finger of a spread hand. The W49 filter was quite fumblingly designed.

(Stockhausen and Kohl, 1996)

(Please listen to Example CD track 10, *Mikrophonie I*. For this example I have taken one side of the stereo recording only to illustrate the filter steps and its characteristic sound.)

This resonates with Jammy’s comments about the Altec filter making “some squawky sounds when you change the frequency” (Veal, 2007, p. 114). Used in this way, the W49 was never going to be a click-free filter as demanded by the score, and it makes it all the more important for this to be taken into consideration by subsequent interpreters. It is also perhaps a little unfair to call the W49 “fumblingly designed” since it is quite clear from the legend printed on the unit itself that it was never meant to be used in this manner.

What this points to is the extent of influence over the final sound of the consequences of repurposing these instruments in the audible results of this process. This can be thought of as a translation of the physical presence of the device into an audible signature, but it can also act via its imperfections as a means by which the physical presence of the performer is revealed, and in this way reinforces the individuality of both the instrument and the instrumentalist, as well as the individuality of each interpretation and performance of *Mikrophonie I*. To refer to the quotation at the beginning of this chapter from the same lecture, the clicks and noises are something to be preserved.

Whether this is post rationalisation, techno-romanticism, or even nostalgia it is hard to say, but if we examine the instrument’s design and construction we can see that the clicks are indeed an inevitable part of its sound just as the key noise belongs to the sound of a saxophone and the hammer noise to the sound of the piano. There is a mismatch here, an inconsistency between the “click-free” specification in the score and the “scraping and skips” being regarded thirtyfive years later as “belonging” to the sound. The noises and imperfections have come to be regarded as signs of fidelity to the work in question. The imperfections are integral components of the original technique. It seems as though the listening practice inherent in performance practice has entrained the composer-performer to accept and incorporate the previously undesirable noises into the composition and identity of the piece itself. The importance of the noise element must be emphasised because there are many interpretations where this has been ignored and where the subsequent results have been adversely affected as a result.

The material nature of the instrument seems to have become more important than the technical specification of the frequency steps by the time he gives this lecture in

1991. The carbon strips referred to by Stockhausen are actually found in the W66c faders and not in the W49 filter which uses sliding metal contacts. Does this betray a lack of technical awareness or is the point he is making more about the use of materials than the accuracy of the specific example? I believe that this comment originates from Stockhausen's adoption of the W49 and W66c setup as his electronic *musical instrument* of choice throughout the period from the mid sixties to the mid seventies. He is possibly referring to the unit as a whole after Gehlhaar had bolted the three components together. By alluding to the catgut strings of violins in the same sentence, he is trying to communicate that this is no mere technical apparatus but is indeed a material musical instrument, with all the idiosyncrasies that come with that status. In other words, he is making a strong case for cultural alignment between the technology and its musical use; the image of using contact spray on the carbon strips is an implied alignment with the use of rosin on the horsehair bow - both actions of material intervention with a common purpose - to improve the sound of the instrument.

Talking to Volker Müller with my electronics engineer's hat on (Müller and Williams, 2011), we had a chuckle about Gehlhaar and Stockhausen using contact cleaner on the controls since we were both aware of how this usually exacerbates any noise problem by attracting dust and grime, even if it may sometimes lead to a temporary improvement. The use of contact cleaner, as well as some of the confused technical details outlined in Section 4.3.3 could be read as the performers and composer betraying a slight lack of familiarity with the technical details, which would add significance to the roles of the various engineers such as Jaap Speck, Werner Scholz and Volker Müller on whom Stockhausen relied during this period.

The affordance of this instrument predisposes it to making clicks and crackles due to the switching in and out of the signal path of inductors and above all, capacitors. This noise also stems from the mechanical nature of the switches but even with liberal use of contact cleaning spray, as described by both Gehlhaar and Stockhausen, the charging and discharging of floating capacitors will still cause all kinds of clicks and thumps and inductors will "kick" regardless of any potential reduction in mechanical switching noise. There is a small amount of acoustic noise from the filter as the faders are clicked through the detented settings, although this would be very difficult to hear by all but the closest audience members, just as the woodwind instrument is partly characterised by locally audible key-noises, the piano by pedal noises and the strings by scraping noises. These are rarely notated as they are invariably considered as less than desirable (if they are considered at all), but as undesirable as they might be, take them away (using audio post-production tools) and the instruments will sound wrong and the music will sound inhuman. There is no reason not to apply the same logic to

the clicks, both electronic and acoustic, of the W49 filter in the context of performing *Mikrophonie I*. Perhaps Zenph Productions (see section 2.4) could deliver a click-free version to oblige the “hi-fi” enthusiasts...

4.4 Performance Practice

4.4.1 Configuration

As shown in Figure 4.1, the W49 had two Maihak W66c volume faders bolted to the left hand side of it, making an integrated performance instrument. The score demands that this whole apparatus is fastened to a small table 40 cm high, and the height difference between front and rear of the W49 ensures that, like the MCI mixing desk, the instrument slopes slightly towards the performer. The combined weight of almost 12 Kg ensures that there would be little movement of the instrument during all but the most animated performance, and the formal concert hall setting with the performers seated in amongst the audience would also have constrained the performers from moving around too much during performances.

The W49 is a passive device and as such requires an additional gain stage to boost the signal back to the original level. The score shows a picture of a Maihak V41 amplifier - a valve design by TFK originating in 1928 - and subsequent photos of performances of *Mikrophonie I* show the ubiquitous V72 - the V41's descendent. It would seem that these are being used as microphone pre-amplifiers between the microphone and the W49, and possibly as gain make-up amplifiers (with a fixed gain of 36 dB) after each W66c, making up the quoted 40 dB loss. Gehlhaar described a unit put together by Werner Scholz made up of eight modular units possibly including V72 pre-amplifiers, with a strong likelihood of the presence of some V74 gain-make-up amps and signal splitters to send the filtered signal post fader to the power amplifiers and loudspeakers. An eight-channel amplifier features in many layout diagrams during the period between 1967 and the late 1970s (Stockhausen, 1972, pp. 30, 99, 117, 131) and I believe this to be the same unit. The microphones used by the ensemble according to Gehlhaar and confirmed by photographic evidence were Neumann KM54 valve condensers. These microphones have a cardioid pattern and require an external power supply. These power supplies can be seen onstage in some photographs, but the location and precise identity of the microphone preamplifiers is as yet unconfirmed.

The means of amplifying the signal after the W49 and faders - the power amplifiers and loudspeakers - are not specified in the score, presumably since this would depend on the size of the performance space and available equipment, however, when taking their

own amplification equipment, the ensemble would use the Altec Lansing loudspeakers from the West Deutsche Rundfunk studios, each containing a Telefunken V69 power amplifier. These active speakers would be calibrated individually and then overall volume control would be from the two sets of W49 and W66c instruments with no need for an intervening front-of-house sound engineer. Whatever colouration these amplifiers and loudspeakers provided would add subtly to any potential noise (such as amplifier hum), but the signature noise of the W49's switched frequency band design is a much more active noise source and therefore has more significance to a study of *Mikrophonie I*. Today, the Stockhausen Stiftung has a preference for Meyersound amplification because of the clarity of sound and the relatively small size of the units which allows them to be flown or stand-mounted higher up than many larger speakers. In each case, the hall in which the performances took place would have had a great influence on the way the piece sounded and although my research does not go into detail about the acoustic qualities of the spaces, the reader should be aware of this nonetheless. For performances in very reverberant halls lower volume levels would have to be used, in line with chamber music levels, but if the hall acoustics allowed, they would make full use of the available speaker power and play loud (Gehlhaar and Williams, 2011).

Since the acoustic sounds are present in the performance space alongside the filtered and amplified sounds the interaction of the two will cause some additional comb filtering and phasing as a result of both time delay between speakers and changing phase response near the cut-off frequencies of the filter. It is vital therefore to consider the final sounds heard by an audience either in the concert hall or at home listening to a recording, as part of a dynamic system - the "inner polyphony within each sound" is heard within the acoustic environment (Stockhausen and Maconie, 1989, pp. 81-2). This mirrors the use by Tubby of filtered sounds in parallel with original sounds, both resulting in phase cancellation effects and a richer sound-world. The impact for the home listener (at least on vinyl) is that filter clicks may be heard as vinyl clicks and therefore "listened through" (Milner, 2009) and not necessarily heard as an integral part of the piece.

4.4.2 Performance Technique

In a lecture delivered in 1991 at the Freiburg Musikhochschule, Stockhausen relates a comment from Thomas Kessler concerning the interface of the W49 filter:

Even if the filter is created electronically with a computer, we should still construct them mechanically like they used to be, no matter what the electronics behind it are. It must still be the same for the hands. (Stockhausen and Kohl, 1996, p. 97)

Stockhausen clearly splits the instrument into its two constituent parts - the sound producing element and the performance interface. We now concentrate on the interface aspect. The high and low frequency limits of the filter are adjusted using two faders in the same slot, therefore ensuring that the minimum bandwidth is always one step apart and the frequencies cannot therefore overlap and completely cut off the sound. That is to say that if the low frequency limit is set at 600 Hz, the high frequency limit cannot physically be set lower than the next step above which is 800 Hz. This allows the performer to pinch both faders between thumb and forefinger and move them both at once, always keeping the frequency bandwidth as narrow as possible but only using the one gesture instead of having to coordinate parallel independent movement. For a wider bandwidth moving in parallel, fingers of the right hand can be placed between the two faders to maintain a consistent gap. This is far more efficient than having two independent faders or knobs which would require the use of both hands to operate. The slope control, in effect a wet/dry mix control is always left at setting 1, i.e. maximum slope, so does not feature in any known performance practice.

Despite these affordances, we still need to understand Stockhausen's accusation - "fumblingly designed." Measuring the force needed to move the faders - 10 N ⁷ - and the distance between the faders at their maximum span - 160 mm - it becomes evident that some parts of the score may be impossible to perform unless the performers have large, strong hands.⁸ This is no surprise with Stockhausen's scores, but worth pointing out in the context of the influence of the instrument on the performance. At the start of the following example which is a detail from another page of the *Mikrophonie I* score (Figure 4.3), the faders must be moved from their maximum distance apart to very close together by the right hand whilst the left hand simultaneously moves the volume faders in fast small movements. There is considerable scope for this to go wrong in performance, or more specifically, for the instrument's affordance to influence the performance practice by changing the composition through necessity of interpretation.

At this point the reader is advised to watch the accompanying video of Rolf Gehlhaar demonstrating some of the performance technique, the salient points being presented in the text below. (See Portfolio DVD *PhD Materials/Rolf W49 Demo.mov*) This demonstration was facilitated by my construction of a mockup of the W49/W66c instrument (see Appendix G).

With the right hand able to govern all filter settings, the left hand is free to manipulate both volume faders; the left fader controlling the rear loudspeaker and the right fader

⁷This measurement was made on number 071 but cannot be taken as anything but a rough estimate for the other units, especially since the passage of time could have had a significant bearing on these results.

⁸Gehlhaar complained of tired and painful hands after thirty minutes of performing *Mikrophonie I*.

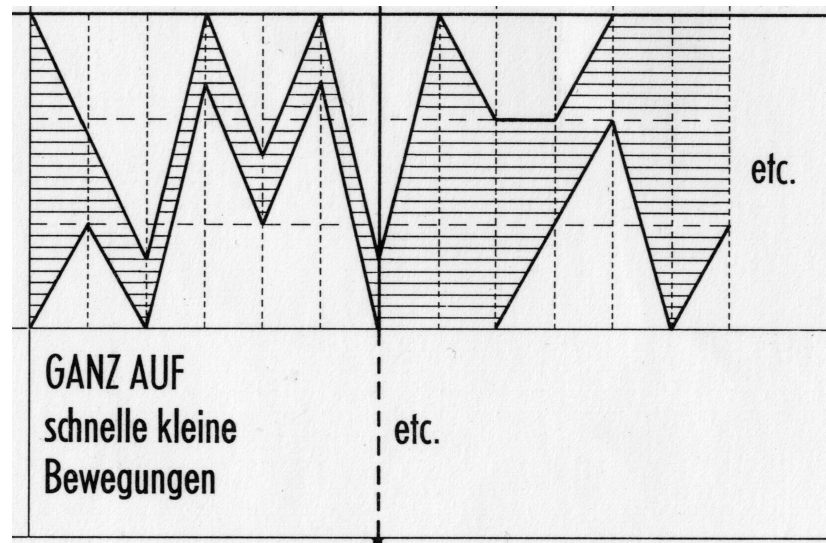


Figure 4.3: *Mikrophonie I* score detail. (Stockhausen, 1974)

controlling the front loudspeaker. By placing the left hand palm up with the left (rear) volume fader between the third and fourth fingers, and the right fader (front) between first and second fingers, both faders could easily be moved, independently and as a unit. Gehlhaar would set up the balance of the loudspeakers so that with the rear fader set at -40 dB and the front fader set at -25 dB, each speaker would be at the same volume level. This arrangement allows the left hand to rest comfortably on the faders at a natural angle with the sound balance also in a neutral position.

The rear fader would be changed by a maximum of between +12 dB and +18 dB during a performance whereas the front fader could be completely lowered and manipulated much more freely as the acoustic sound of the tam-tam in front of the audience would always be present. The sound could be made to flow by manipulation of the front fader, leaving the rear fader static, either just by twisting the hand with the existing grip, or by adopting a slightly different grip, using the thumb and forefinger to adjust the front fader. Gehlhaar cites the fact that people face forwards as the reason for not adjusting the rear fader very much, preferring to use the front fader to chop the sound and to give greater expression. With the faders this way round the more sensitive thumb and forefinger can control the more critical front loudspeaker.

The notation for the sound projectionist only addresses the overall amplitude of the signal so the performer is free to make front and rear adjustments intuitively. It is clear that a different configuration, for example one overall volume fader and a pan-pot (as used by Burns (Burns, 2001)) or even a crossfader governing front and rear distribution would lead to a very different technique and much less ergonomic solution for sound projection. With the two independent faders, the sound projectionist is free to shape

any crossfade between front and rear, and is able to exercise great influence with a minimum of controls, not constricted by the automatic adoption of the mixing desk paradigm so prevalent today.

Gehlhaar also demonstrated some techniques of reducing the audibility of the clicks by quickly lowering the volume fader, adjusting the filter frequency and raising the volume fader again. In conjunction with the acoustic sound of the tam-tam being present in the space simultaneously, this, Gehlhaar said, would result in a little bit of amplitude modulation or tremelo instead of a click, but he was careful to add that sometimes the clicks were actually useful or appropriate, so this was an aesthetic choice born of practice. He also drew attention to the fact that, although the clicks were somewhat unpredictable, they were much more likely and much more noticeable (and possibly stronger) when filtering lower frequency material such as when the microphone picked up sounds from the edge of the tam-tam (Gehlhaar and Williams, 2011). The use of a cardioid microphone in close proximity to the tam-tam would certainly result in emphasised low frequency response.

Gehlhaar's account of performances is particularly enlightening in relation to the affordance of this instrument. He reports difficulties in seeing the on-stage performers and following exactly where they were according to the score. This forced him to improvise and he was afraid of Stockhausen noticing this until after a few performances he noticed that Stockhausen was having similar problems himself and was also having to improvise. Recall Bunny Lee's comment about Tubby's multiple mixes of the same song: "And of course, every time it would be different" (Bradley, 2000, p. 316). In this case, performances would be different, but constrained by the affordance of the instrument and the practice developed through many performances of the same piece.

The three filters in Figure 4.4 all show the two thumbscrews to the left hand side which were added to allow the W66c faders to be attached as in the leftmost unit. This connects these filters with use by Stockhausen. The filter in the centre which unfortunately has no serial number, is the most worn, and possibly the oldest of the five W49s at the WDR Studios. The worn away paint at the lower right hand side strongly suggests a pattern of usage aligned with performance rather than a set and forget style, and its location suggests that it was caused by the heel of the right hand which would have rested there between fader movements. The slider controls on this unit were in the worst condition of all, also suggestive of extensive use, possibly as Stockhausen's favourite W49.



Figure 4.4: W49 filters at the WDR Studio.

4.5 Comparison of Practices

4.5.1 *Mikrophonie* / Interpretations

After a performance of the piece given in Kürten in 2010 by the *anthos Ensemble* using a different filter, without steps or clicks, Kathinka Pasveer noted that the sound was too clean and that Stockhausen preferred it when the filters were crackly and “kaput.” The filters used by the *anthos Ensemble* (see Figure 4.5) were purpose built by Jan Panis and feature on-board microphone pre-amps with gain control pot (wired back-to-front), two faders for 2-channel output gain and two *side-by-side* faders for high-pass and low-pass filter frequency. There is a metal spike that prevents these faders from passing one another, but the fader caps are standard style and not the upright paddle type found on the W49 (originally from the W44 fader). The filter slope seemed very shallow and there was little resonance to them either, making the filtered sound lack bite. The smooth frequency change and lack of steps robs the sound of aggressiveness and makes the piece sound more polite than the recordings made by the Stockhausen Ensemble (Karlheinz Stockhausen, 1966, 1975).



Figure 4.5: Panis filter used by *anthos Ensemble*.

Watching both the rehearsals and the performance of the *anthos Ensemble*, it was noticeable that having access to the microphone pre-amplifier control meant that this was adjusted during the performance, perhaps distracting the attention from the sound projection itself. The LED meter was also a distraction, with a lot of adjustment to the microphone preamp level in conjunction with observation of the LEDs.

Burns describes a realisation of *Mikrophonie I* in which the filters were created in Max/MSP (Burns, 2001). He notes the importance of the stepped nature of the filter and models this, and by iterative testing in rehearsal picks a steep slope for, as we understand, aesthetic reasons. However, without resources for constructing a hardware control, he relies on standard MIDI faders and runs into the problem of how to deal with the overlap of frequency selection. His solution is to setup the Max patch to prevent any overlap, forcing the overlapped frequency to move along with the altered frequency, maintaining the smallest filter bandwidth, but thus decoupling one fader in the process. Panis' simple but effective mechanical solution - a pin physically preventing the side-by-side faders from travelling past one another - could have been employed here, and this

would have maintained the relationship between fader position and filter frequency at all times.

In the light of the research presented in this chapter, one of Burns' more contentious decisions is to change the configuration of the front and rear volume faders for a single volume fader and a panpot. This makes some of Gehlhaar's techniques - ducking the front volume briefly whilst changing filter frequency and leaving the rear volume unchanged - impossible, and disrupts the relationship between volume and spatial position that is an inherent characteristic of the dual volume fader layout. It also makes it much harder to control all parameters with just two hands, and transforms the approach to the front/rear balance, aligning the practice more with a standard mixing desk model than the individual *Mikrophonie I* model.

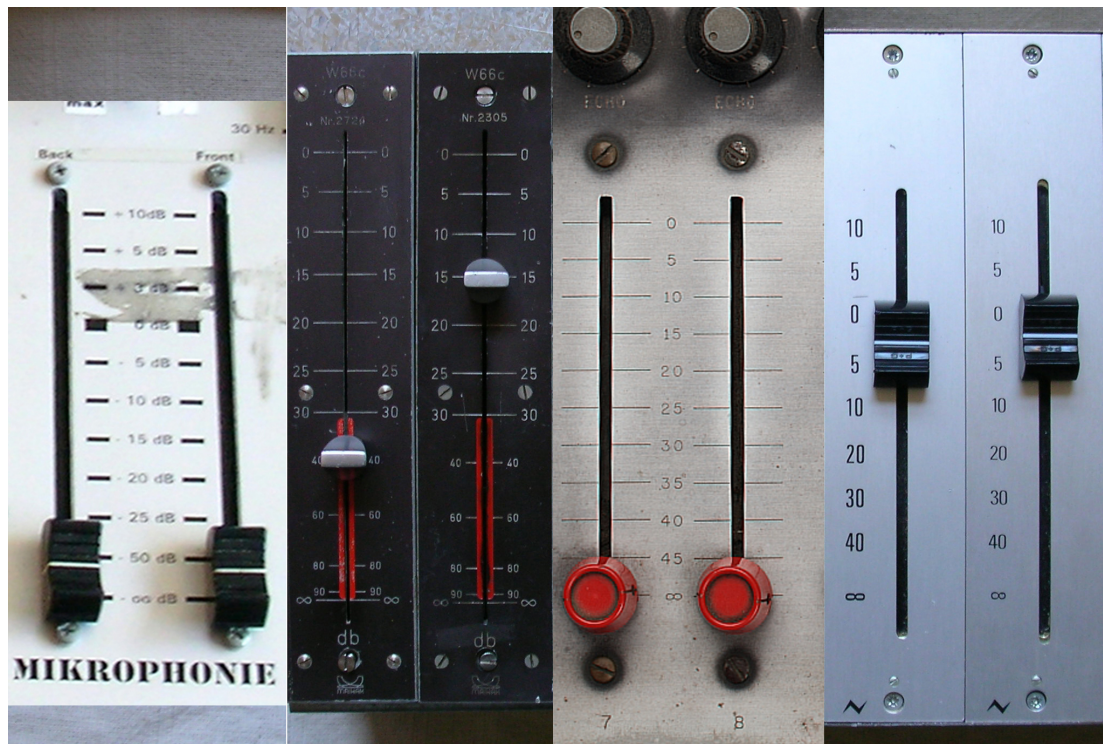


Figure 4.6: Fader Scales. L-R: Panis, Maihak W66c, MCI (Langevin), Neve (Penny & Giles).

In Figure 4.6 I have arranged photographs of four different faders (at the same scale) in order to observe critical performance details. The gain-scaling figures on the faders should be taken as relative dB values as the absolute gain value depends on the way the fader is wired into the circuit. The usual configuration is to have 10-15 dB of gain make-up available. This is observed in the legend on the Panis and Neve faders (far left and far right). The Maihak W66c fader (second from left) was usually used in the studio with a V74 gain-makeup amplifier/buffer supplying 15 dB of gain (Müller and Williams, 2011), so the net 0 dB position would be at the position of -15 dB on

the fader, thus allowing an extra 15 dB of boost if needed. The gain-makeup of King Tubby's MCI desk (Langevin faders second from right) is unknown.

The first detail to note is that the Neve⁹, Panis and MCI faders have a somewhat standardised 100 mm travel whereas the older Maihak fader has a 130 mm travel. This greater fader length affords greater subtlety and precision of action, especially when taking into account the scaling of the gain response. We can see the differences laid out in Figure 4.7. What is immediately obvious is that the breakpoints in the response curves occur between 35 dB and 50 dB below maximum, and although there is some slight variation above -35 dB (particularly with the Neve scaling), the curves are very similar in this upper range. It appears that the extra 30 mm of travel afforded by the W66c fader (as used by Stockhausen) is employed in the range below -40 dB. If we remember that Gehlhaar would set the rear fader at -40 dB with the front at -25 dB to achieve a balanced signal, that still leaves 40 mm of fader travel to attenuate the rear signal (i.e. plenty). Using the Neve fader in this amplitude range, you would have only 25 mm of travel but you would also have to negotiate a breakpoint only 5 mm from the end of the fader travel. The short distance (20 mm) between -35 dB and infinite gain reduction exhibited by the Langevin fader as featured on King Tubby's MCI mixing desk, aligns their functionality more towards being used as quite effective on/off controls rather than gradual slow fade devices. The equivalent attenuation afforded by the Maihak W66c would need 50 mm of travel, half the entire length of the Langevin fader, and so allows much more subtle low level amplitude control.

Communicating such details graphically is a poor substitute for practical experimentation but these details are included here to emphasise the importance of the different performance responses associated with these different faders. It should be obvious that substituting Maihak for Langevin faders in interpretations using either Stockhausen's or Tubby's techniques would demand very different techniques to achieve consistent results, and taking into account the shape of the fader caps, it becomes apparent that the physical performance techniques are incompatible. The caps of the Maihak with their narrow, paddle-like shape and additional height allow two faders to be gripped easily between the fingers. The same dual grip is not possible with the round, larger diameter and lower profile 'Rolo' style caps of the Langevin. Gehlhaar's performance technique is therefore not transferrable. Even if modification were made to replace the caps, the different fader curves would present more problems, although at least re-mapping would be possible with a digital system.

Another glaring difference is the audible result on the signal of substituting several valve amplification stages - microphone, preamplifier, gain make-up, and power amplifier - with solid state and surface mount technology. The tam-tam's huge dynamic range is

⁹The Neve fader module has a Penny & Giles fader with 104mm travel.

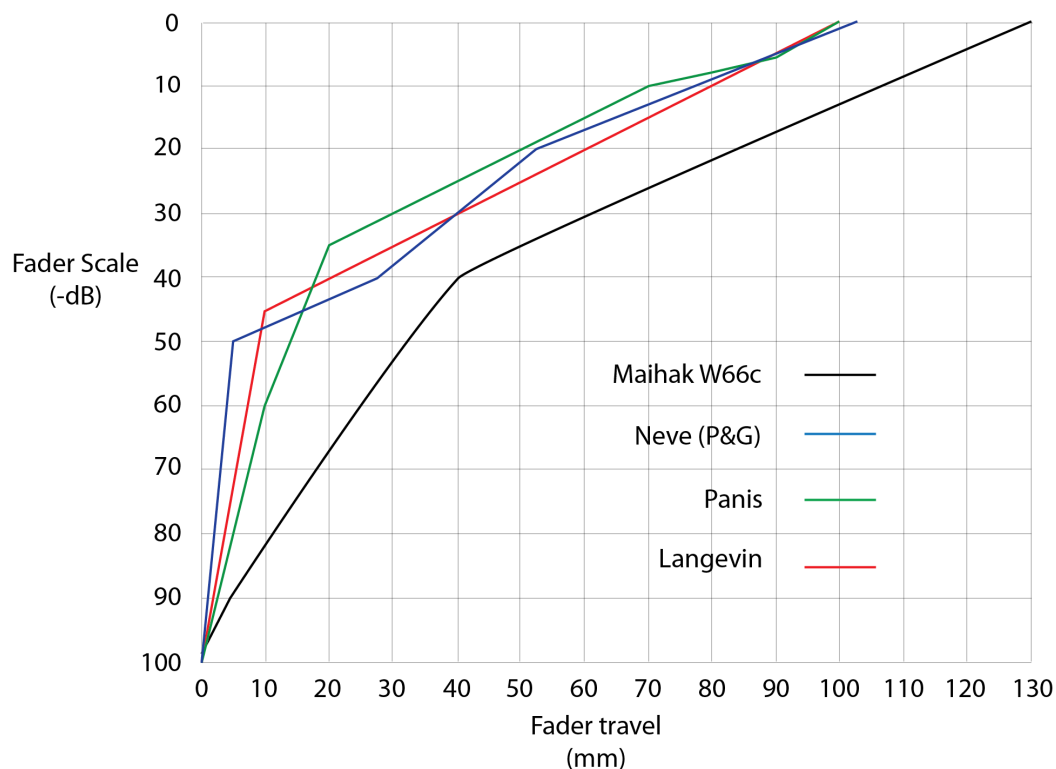


Figure 4.7: Fader Curves

tamed through second harmonic distortion in a valve signal chain, but the distortion in a valve-less signal chain is likely to be far less musical. A valve signal chain will naturally compress the signal (reducing dynamic range), and the effects of such a setup should be considered by contemporary interpreters before substituting these elements with less forgiving devices. The use of compression is permitted by the score yet this was not employed in the *anthos Ensemble* interpretation.

The filter and potentiometers are acting as part of a polyphonic whole alongside the tam-tam playing and microphone performance and this last item is as important for creating an accurate and faithful sound as the other two elements. The score directs the microphonist to hold the microphone at close, medium and far distances from the site of excitation in two ways; parallel to the surface of the tam-tam; perpendicular the surface of the tam-tam. The first part is self-explanatory however the critical information missing from the second part of the explanation is how far away is “far?” The *anthos Ensemble* held the microphones a metre or more from the tam-tam’s surface at times but the problem with this is that almost no signal reaches the filters and therefore the loudspeakers, so the effect of the electronics is drastically reduced.

Gehlhaar described how he would hold the microphone around six inches away from the surface for the sections notated as far away, and sometimes only four inches; two to three inches for the medium distance; and right up close without actually touching the surface for the close sections (Gehlhaar and Williams, 2011). (See Appendix E.2.2).

Stockhausen describes his early experiments and a similar distance from the surface of the tam-tam is mentioned:

This tam-tam was hanging in my garden: I couldn't put it in the living room, it was too large. Every once in a while, when I went out for a walk in the garden, I would take a pen or a key, and scratch it, or just knock it with my finger, bang it with a pebble, write on it with the pebble, and then often lean my ear very close to the surface of the tam-tam, where I would hear all sorts of strange sound vibrations. At a distance of four or five inches away from the surface, these sounds were no longer audible. (Stockhausen and Maconie, 1989, p.77)

Because of the extreme microphone distances (a metre or more), the electronics performers in the *anthos Ensemble* struggled to control microphone gain at the same time as operating fader levels and filter frequencies. Perhaps the inclusion of the microphone gain control on the Panis device contributed to causing this problem by transforming the question of microphone practice into one of sound engineering. This technical feature aligned with a tendency to assume electronics performers are also sound engineers or technicians whose job it is to make everything work for the acoustic instrumentalists placed the filter players under considerable pressure, and prevented the microphonists from developing a more robust practice taking responsibility for understanding how the microphones respond. Gehlhaar thought it obvious that as well as following the score in terms of microphone distance, if there was a very loud sound required such as slamming chains against the tam-tam, the microphonist would naturally increase the microphone distance to compensate - a natural compression effect. This shows the importance of developing this practice within a wider "performance ecosystem" (Waters, 2007), maintaining a practical awareness of the synergetic relationship between the many subsystems involved.

Taken in conjunction with other directions in the *Mikrophonie I* score to "ad lib.", the physical nature of the filter and the way in which performers necessarily interact with it become vital to the successful/faithful communication of *Mikrophonie I* to an audience, but such improvisation must be done sympathetically with relation to the system as a whole comprising all three members of each group. This is perhaps where the techniques of electronic music performance, many of which overlap with a technical engineering practice, can be of the utmost importance. The *anthos Ensemble* are principally percussionists of the greatest ability, and for performances of *Mikrophonie*

I they alternate between roles. It seems as though the electronic practice is not necessarily being seen as demanding the same levels of skills as the percussion practice, but this is exactly the sort of area which my research hopes to address. Stockhausen's score gives instructions to the electronics performers but we only realise how much detail is missing and therefore how much performance practice is undocumented when comparing different realisations and performances. Technical expertise within a musical context is absolutely essential for a successful interpretation of this piece, and it is only with the right alignment between technology, design and practice that this can happen.

4.5.2 Tubby and Stockhausen

Several details throughout the last two chapters should have supported the case for a similarity of practice and I now summarise the main points and add comment and detail where necessary.

Making a technical comparison of the filter frequency specifications of the Altec 9069b high-pass filter and the high-pass section of the W49 filter, it is possible to see the similarities in terms of the number of steps available, the range, and the relative frequency values in the two filters, both contributing to their similar sonic characteristics. Figure 4.8 shows a plot of the frequency steps in each filter.

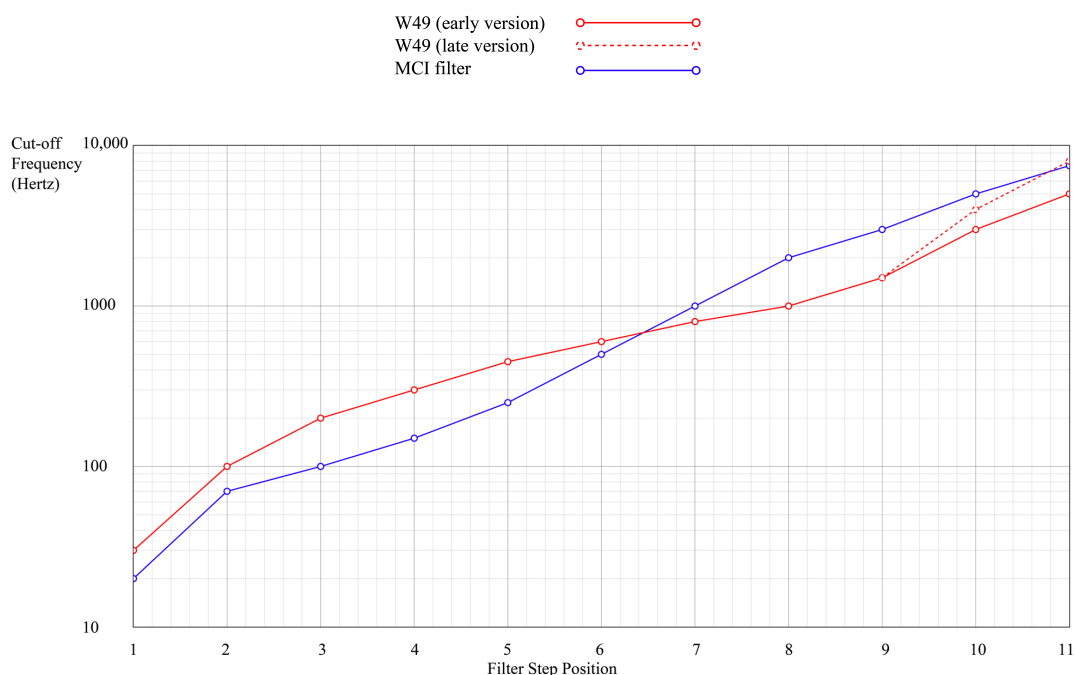


Figure 4.8: Filter frequency comparison chart

A noticeable difference is the greater definition in the low-mid range of frequencies of the W49. Between 100 Hz and 1 KHz, the W49 has seven steps whereas the Altec filter has only 5 steps. This means that the W49 filter has better resolution throughout the mid-range where the Altec filter has slightly more control at the extreme ends of the frequency spectrum, and this is evident in many of Tubby's mixes, especially at the higher end of the frequency spectrum. It must be noted that it has not been possible to take any frequency measurements of these devices, so the data represented above remains unconfirmed beyond the manufacturers' specifications.

Both filters are based on a similar, although not identical configuration of two inductors and one capacitor, with all components being switched simultaneously to obtain different frequency settings. This generates a similar type of switching noise, based mainly on the nature of the electrical response through the components, and additionally on the mechanical contacts of the switching mechanisms.

Whilst the W49 has a linear frequency slider switch, the Altec 9069b has a rotary switch, however both controls offer the ability to select the entire frequency range in one relatively easy movement of the hand.

Both filters were repurposed by their users from static set-and-forget type effects filters into dynamic musical instruments, and were incorporated into a wider musical practice also reliant on repurposing and extension of the normal boundaries of technology and technique. Indeed, it is striking that Tubby's and Stockhausen's repurposing of their respective filters sprang from improvisation in both cases. One of the main features of instrumental improvisation is a tendency to employ extended techniques, coaxing unexpected or non-standard sounds from existing musical instruments. Tubby's manipulation of the metal spring from the spring reverb unit is comparable to Stockhausen's use of the tam-tam as a resonant metal plate, with both devices responding to being bashed or shouted at in a similar way.

Both practices share an approach to configuring the instruments so as to allow an economy of movement to control a wide range of audible effects; Tubby's one-fader control over delay level and feedback compares to Stockhausen's one-handed control over both filter frequency faders or Gehlhaar's one-handed control over both volume faders.

Several sections in Stockhausen's *Mikrophonie I* score call for the sound projectionist to improvise filter and fader movements, and all the choices about how to distribute the sound between front and rear loudspeaker are left to the discretion of the sound projectionist. If we consider Gehlhaar, and Stockhausen's other assistants being in a similar situation as Jammy, Scientist, and Tubby's other assistants, there is therefore a case for arguing that some stylistic influence was provided by Stockhausen and Tubby,

guiding the improvisations. In Tubby's case it could be described as a house style emerging from his initial experiments, and in Stockhausen's case we could think of each piece having its own identity, comprised by the score and the associated practice, especially the tacit knowledge of the performers. A Stockhausen W49 style is harder to codify since, except for Gehlhaar, Johnson and Maiguashca in Osaka (Gehlhaar and Williams, 2011), and the need for a second W49 performer in *Mikrophonie I*, he rarely seemed to allow others to take his role, and therefore did not need to annotate or otherwise document his own technique. Some general principles might be understood through a listening analysis of recordings, however, the sort of filtering settings used are very hard to abstract from the huge range of acoustic sounds generated by Alings and Gehlhaar on the tam-tam and Fritsch on the viola.

In order to make a comparison of the performance technique between Stockhausen's use of the W49 and Tubby's use of the Altec I have transcribed a mix of *Rebel Dance* which features use of reverberation and the high-pass filter. I am not suggesting that Tubby worked from a score, but the intention is to compare the performance on the stepped filter to that indicated in the *Mikrophonie I* score and ultimately to use a recreation of the filter to explore further the performance practice involved. Figure 4.9 is a detail of a page from *Mikrophonie I*.

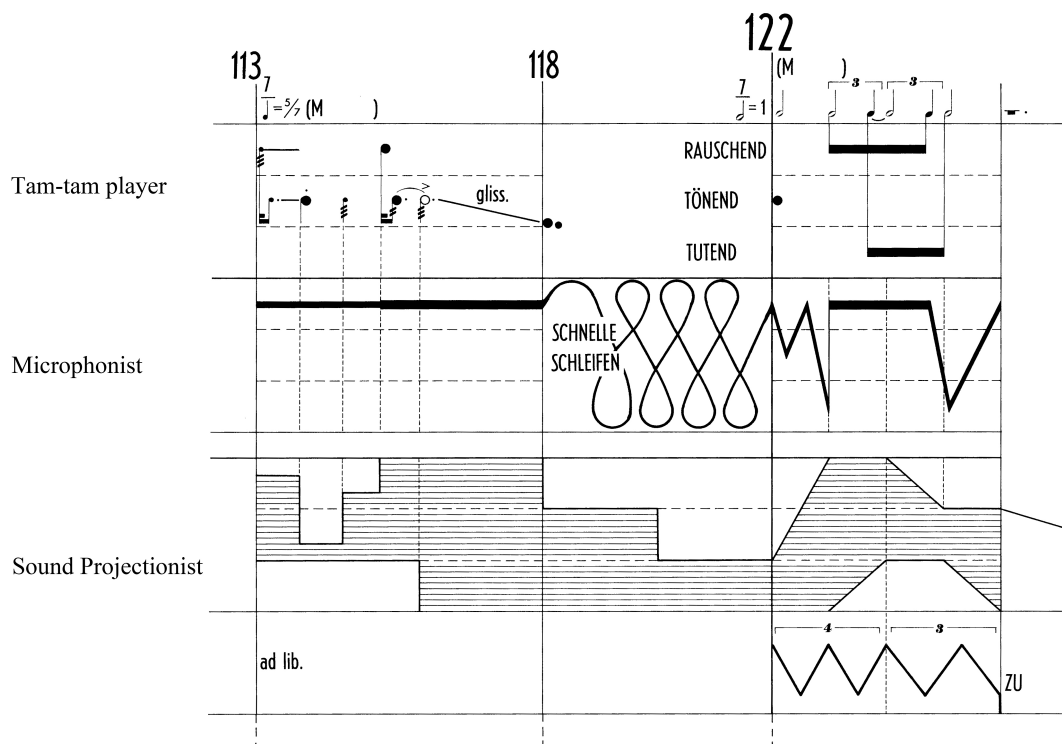


Figure 4.9: *Mikrophonie I* excerpt [my labels to the left] (Stockhausen, 1974)

This notation is for three performers; the tam-tam player follows the top line; the

microphonist follows the middle section and is directed to hold the microphone near or far from the surface of the tam-tam and also near or far from the site of excitation; the sound projectionist follows the lower section, the shaded area representing the frequencies allowed to pass through the band-pass filter, and the line in the bottom section representing overall amplitude as governed by the W66c potentiometers attached to the left hand side of the W49 filter (see Figure 4.1).

Concentrating on the filter notation, the two features most notable in this excerpt are the clear steps from measure 113 to 122, and the notated slopes from measure 122 to the end. This points to a similar usage of the filter as Tubby's for both sweeps and rhythmic punctuations, again using the stepped nature of the filter for rhythmic musical effect. The tie-lines between the filter steps near the bottom of the page and the tam-tam events on the top row are testament to this, the filter frequency steps being exactly coordinated with strikes of the tam-tam, not preceding or following them.

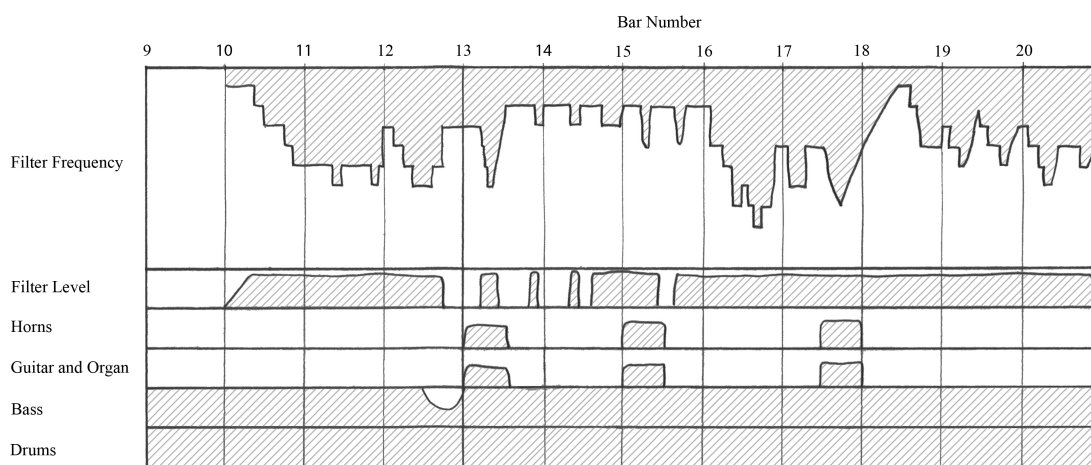


Figure 4.10: Transcription of mixing desk performance

In the transcription of *Rebel Dance* shown in Figure 4.10, I have deliberately emulated Stockhausen's sound projectionist notation. Instead of transcribing the notes played by drums, bass, guitar, organ and horns I have simply included bar numbers and indicated the amount of each instrument in the mix as governed by the volume faders on the mixing desk. Since the mixing was done from four-track tape, guitar and organ were grouped together on one track. This also reflects the fact that no instruments were present at the time of mixing - that part of the performance in effect being done by the tape machine. With bar numbers read from left to right, the first row shows the filter frequency with the shaded area representing the frequencies allowed to pass through the filter. Note the lack of an upper frequency limit due to this filter not having a low-pass element. The second row shows the amplitude of the filtered signal just like the *Mikrophonie I* score, and the next four rows show the amplitude of each

of the four tape tracks. Reverberation is applied individually to each tape return but notation for this has not been included here.

(Please listen to Example CD track 11, King Tubby: *Rebel Dance*, which matches the notated excerpt.)

At measure 16 there is clear use of the discrete filter frequency steps to create a triplet punctuation whereas in many other parts of the mix the filter is perceived as sweeping in a more continuous fashion. This is an illustration of the steps in the filter and the related transients being used to impart additional rhythmic elements to the underlying sound, just as is evident in many places in Stockhausen's score. Indeed, Stockhausen uses stepped notation as well as slopes to imply this very distinction. Stockhausen, of course, specifies the usage of the filter in the score but by his own admission, the score is the formalised result of a series of experiments he performed with the tam-tam in his garden and the filter in his kitchen.

Tubby's use of the faders is more for bringing out tracks at particular beats or, as we can see here during measures 13, 15 and 17 for injecting a blast of horns and rhythm. Stockhausen's fader use is not at all related to any repetitive rhythm, but is much more gestural and is related more to each particular moment, seeking to augment the acoustic sound by reshaping its envelope or imparting a new dynamic shape to its internal rhythm.

4.6 Conclusion

Although this research stemmed from my perception of sonic similarities in the work of King Tubby and Karlheinz Stockhausen, it has been possible to trace certain similarities at the heart of both of their approaches to making dynamic musical instruments out of otherwise static technical devices. The striking factors are the way in which noises and environmental sounds, often undesirable from a compositional point of view, have become part of the authentic sound despite the initial struggle against them. To my mind, these imperfections end up by emphasising the physicality of the performance and therefore the material nature of the music making process, and so allow us, as listeners, a closer appreciation of the physical origin of this music. This performance aspect which lies at the root of both of these very different music makers' approaches is a vital part of what links their music making activities in the absence of any other connection between them. It reaches across geographical, social, commercial and economic boundaries, which otherwise position their work at opposite ends of the spectrum, and sheds light on a common musical purpose through the joy of the musical manipulation of sounds using any available tools.

It should, by now, be obvious that these kinds of electronic devices, whether used within their original operational parameters or extended and repurposed in the ways detailed above, must be considered as individual musical instruments and not simply as manufactured electronic tools. Their use as instruments relies on the relationship between their physical and electronic design, and bound up with this is the interface that they offer the performer and the relationship between the physical and aural feedback the performer experiences.

The duality of technological resources and the composition process (Manning, 1999) are combined so well in these two examples, and rather than being antagonistic, the two elements work in great harmony. It is the performance practice, the playing of the instruments, even, as in Tubby's case, at the production stage, that transcends the merely technical and joins up the design with the composition. This implies that a removal or substitution of one or both elements would make it very difficult to create an authentic analogue to this practice. Noise, incidental or otherwise, plays a major role both in the sound of this music and the way we have been able to analyse it.

Gehlhaar's account of the improvisation that went on during performances of *Mikrophonie I* reinforces the live nature of this electronic music, but without Stockhausen or Gehlhaar to do the improvising, the technical side of this duality is perhaps even more important in its ability to influence, but not to *determine* the outcome. On its own the technology cannot make a performance sound faithful to Stockhausen's conception, and on their own, without the appropriate technology (as we saw with *anthos Ensemble's* performance in 2010) skilled musicians cannot necessarily do that either. What is needed is the alignment of clear technical expertise with the right instruments and the right performance skills, all of which are equally important.

By examining the mechanisms by which these two instruments have shaped the music, and by observing the physical interaction between performer and instrument, we can understand how a particular technology can have a two-way relationship with the composition, both shaping it and being shaped by it. Beyond that, a material study enables us to re-evaluate such instruments and, with the benefit of hindsight, to discover certain characteristics, like Stockhausen's subsequent appreciation of the "scraping and the skips between filter-levels" (Stockhausen and Kohl, 1996), which may have been struggled with at the time, but have since been recognised as essential to the nature of the instruments and the music made with them. It is so often the case that a struggle to overcome certain limitations, here exemplified by the repurposing of both of these devices, yields far more interesting results than a situation where everything is too easy, where the tools are used according to the manufacturer's prescription. That kind of situation runs the risk of technological determinism, or even worse, corporate determinism, and must, as Attali proposes (Attali, 1985), be resisted at any cost.

As a musical instrument, the stepped filter is like an evolutionary dead-end in that examples designed after the nineteen-sixties are almost impossible to find outside laboratory inventories. In recorded music it only really features in repurposed form. Stepped high-pass filters are often used in mastering studios or on some mixing desks, but I am unaware of other physical examples exhibiting such a wide range of cutoff frequencies and being used by dynamically changing the frequencies. Improvements in electronic component specifications, voltage control systems and eventually digital control eliminated the need for limiting musical instrument type filters to discrete steps, so subsequent designs used in synthesizers or effects units featured continuous control instead. It is easy to understand how changing attitudes towards noise, progress in design, and even the value of vintage equipment make a thorough examination so important.

In these two examples we have a differing amount of technical mastery. King Tubby is first an electronics engineer and designer, and his knowledge and technique grow from this side of the continuum through performance practice and into composition. Of his protégés, Scientist is the more technically minded whilst Prince Jammy is less technical and perhaps more biased towards the performance practice. Stockhausen, on the other hand is principally a composer but with extensive performance practice experience and a good deal of technical knowledge. However, he has the luxury of the technical and financial support of a massive broadcast corporation and so has engineers; Jaap Speck, Werner Scholz, Volker Müller and others, to come up with creative technical solutions to his problems. With Stockhausen's presence looming so large, it can be difficult to recognise and appreciate the contributions of these technicians to the overall sonic output, but as I have tried to show, they have a significant influence over the culture/technology alignment that allowed such music to be produced.

In 2012, aside from special situations such as STEIM, it is rare to have the same level of technical support to design and build performance instruments to specific requirements, so there are two options for the composer or interpreter of electronic music. She can rely on mass produced commercial hardware and software and try and adapt her composition and performance practice to these kind of devices, possibly using software modelling to emulate electronic instruments, or she can empower herself by learning how to design, build, or modify electronic instruments that will be suitable for the task in hand. Even if all the technical skills to make one's own instruments cannot be fully developed, any progress along this trajectory cannot but help to inform any performance practice in a positive way and will also have an influence on how pieces are composed for performance.

Chapter 5

Practice: Performance of *Spiral*

A recording of my performance of *Spiral* can be found on the Portfolio CD track 7. The Quadraphonic Panner used for sound projection and for mixing the recording is documented in Appendix A.

5.1 Research Questions and Methodology

The material research set out in the previous two chapters forms the basis for experimentation into the technical and performance strategies needed for an authentic interpretation of a piece of electronic music from the 1960s. Choices of solo pieces for electronics by Stockhausen are limited, and since *Solo* (Stockhausen, 1969a) deals much more with specifically delay based technology, *Spiral* (Stockhausen, 1973) was the natural choice.

In this portfolio performance piece my aim was to examine related performance practice issues by limiting myself to using the kinds of technology that would have been available at the time it was composed. These limitations were also intended to aid the production of an authentic, or high fidelity performance. Two performances have taken place and working from this solid foundation the piece is under constant development with a view to being performed at the Stockhausen Courses 2012 in Kürten, Germany. Newer technologies are being added to the setup as particular techniques and responses to the score are extended.

5.2 *Spiral für Einen Solisten*

Written in 1969 *Spiral* is a structured improvisation using +/- notation for a soloist playing any instrument. Two notable recordings (Karlheinz Stockhausen, 2009), (Karlheinz Stockhausen, 1994) feature versions by Harald Boje and Péter Eötvös playing electronic instruments, so my interpretation using synthesizer is consistent with historic practice. *Spiral* engages with the *reproduction stage* (see Chapter 2.3) in its use of shortwave radio which the soloist must use to find and create events from broadcast music, speech and interference, which must then be transformed with the instrument. This is a direct inclusion of sounds transduced from the electromagnetic spectrum which surround us all the time and *Spiral* therefore relies quite heavily on the characteristics of the sound that is received and the process of reception.

Spiral was written for the 1970 World's Fair in Osaka, to be performed in the spherical auditorium of the West German Pavilion. Although there are no specific notated spatialisation instructions as in the closely related *Pole* (for two) and *Expo* (for three players), a sound projectionist is required for performances of *Spiral*. In discussing the reception of the piece with Peter Nelson, he drew attention to the dual nature of its operation in two particular areas; as a piece for two soloists - instrumentalist and sound projectionist - not one; and as an exploration of the boundary conditions between two worlds, embodied and disembodied. Being able to engage technically and creatively with the spatialisation fully was therefore a priority.

5.3 Instruments

5.3.1 Shortwave Radio

On studying the score for *Spiral* as well as for *Kurzwellen* and especially after a discussion with Suzanne Stephens¹, it became clear that the biggest technical challenge was the reception of shortwave signals. The shortwave aspect is one which strongly links this and other similar pieces from this period of Stockhausen's music to the key issues in my research: noise, environment and the repurposing of machines, and it also asks urgent questions about strategies for maintaining and developing a performance repertoire of electronic music, beyond even the conservation of standalone instruments such as synthesizers. If, as Simon Emmerson proposes,² the sound of radio as a medium

¹Stephens learned *Spiral* under Stockhausen's direction in the 1970s and continues to teach Stockhausen's music and practice through the Stockhausen Stiftung of which she is a director.

²Emmerson raised this in a question asked after I presented my *Stockhausen Meets King Tubby* paper at the ICMSN 2009.

is in itself central to Stockhausen's composition in this period, this only emphasises the importance of ensuring good shortwave reception.

Photographs of the Stockhausen Ensemble in the 1960s and 1970s show the use by most players of the Telefunken Bajazzo TS 201 radio with inbuilt aerial extended. I purchased such a radio from Germany, but after trying at different times of day and with different but admittedly basic aerial configurations I was unable to get sufficient reception of very much at all. It transpired that my radio had only the Marine band, which I was led to believe now carries little traffic, but coupled with the insufficient length of aerial and the massive amounts of noise swamping the shortwave bands from all the micro-processors in broadband routers, computers, toasters, microwaves etc. I was at a great disadvantage. The authentic equipment no longer functioned in the same way due to its synergetic relationship with the environment.

Whilst the score allows for the use of other bands in extreme circumstances (i.e. Medium Wave, Long Wave, and FM), the noisier sounds typically associated with shortwave such as single-sideband-modulation, code stations and other more specialist traffic are sonically important. It is inadequate to rely only on the use of music and speech programmes for a performance of *Spiral* as the piece offers a wonderful opportunity for the performer to tap into and respond to the wealth of natural and artificial signals and electromagnetic waveforms.

The difference in the quality and quantity of traffic between the 1960s at the height of the Cold War, and the 2010s is significant, but experiments with a Realistic DX-300 yielded far better results. The DX-300 is more of an enthusiast's receiver, not as portable as the Telefunken but far more precise in its tuning capability and covering the entire range of shortwave bands. This still relied on connection to an external aerial which I constructed using two eight metre lengths of wire, connected in the middle to a coaxial cable which could then be run to the receiver. The dipole design is a simple but effective type, and the length was chosen to be practical to erect and to offer a reasonable chance of receiving signals on the 31, 22, and 21 metre bands (Noll, 1984, p.18).³

I had tried the new aerial with the old Telefunken Bajazzo radio in the Reid School of Music, at home, and in the back garden, and received almost nothing except noise. On testing this configuration in the middle of a green open space⁴ well away from any buildings and sources of radio frequency interference (RFI) I was easily able to pick up a whole range of music and spoken word stations in Russian, French and Spanish. This quality of reception was impossible to translate to the concert hall, but it restored my

³Steve Harris from On The Air in Chester has provided useful advice with respect to how to setup an aerial and which bands to search at what times of day.

⁴The Meadows in the centre of Edinburgh.

faith in the design of the Telefunken radio and its suitability for use by the Stockhausen Ensemble, operating in an environment free from much of the RFI we have today.

Since the experiences with these two different radios I have now settled on using a small “World-band” receiver - a Roberts R871 - which has twelve bands of shortwave and picks up a fantastic amount of traffic even using its own built-in aerial, as well as having a minijack output socket, (useful for volume automation) and immediacy of tuning control. This is only about ten years old but it is much better suited to receiving and producing the more useable shortwave sound than the more authentic period instrument - the Telefunken Bajazzo - and its control interface can be much more readily repurposed for use as a musical instrument than the more professional DX-300 despite the DX-300’s superior tuning precision. It is also pocket sized and has the added advantage of making my setup more portable.

5.3.2 Synthesizer

It is essential to keep the parameters separately accessible during synthesis. This is the crucial point. [...] I do not mind if the parameters are simulated rather than real: dynamics, pitches, and timbres are what I think in, and I want to be able to continue to influence them separately until the result is completely finished. I would like even more parameters, for instance to control degrees of density and aleatoric distribution of musical events within defined limits. In fact, everything should be parametric. Meaning, there should be a button for it.

(Stockhausen and Maconie, 1989, p.134)

My own practice has developed around the constant requirement of instantly accessible parameters with the finest possible control. For this reason I use analogue electronics. My interface with my machines is predominantly tactile, consisting of the usual knobs and switches as well as a Doepfer ribbon controller offering both position and pressure sensing, and three footpedals offering continuous (i.e. not stepped) voltage output and/or voltage and signal attenuation. Stockhausen recognized the benefits of foot control as well as low latency in a wish list for the new WDR studio in conversation with Robin Maconie (Stockhausen and Maconie, 1989, pp.132-3), and having immediate control of each parameter is vital in enabling the quick responses necessary for performing this, and other pieces. MIDI control is unsatisfactory because of its 127 step limitation which often prevents the exploration of the most interesting boundary conditions, in many cases is translated into clearly audible and undesirable steps in parameter changes, and also frequently makes it impossible to home in accurately on the perfect sound for any particular moment. Consequently I use either direct

parameter control via potentiometers, or control voltages which can be adjusted at the tiniest level to achieve the desired results.

Because of the need to emulate sound events, and especially to transform pitch intervals of such events I needed a way of being able to control the frequency of my oscillators selectively according to a western 12 tone scale as well as continuously. For this I adapted a self-built module by adding a switchable bypass function so that a voltage quantizer could be switched in and out of the control signal path. Whilst doing this I added a switch to bypass a slew limiter, and took the opportunity to make some normalised connections to the module to reduce the need for patch cables but also to allow easier access to the switches during performance. (See Appendix B.4).

The use of footpedals has always been a part of my practice as a logical extension of simultaneous control over a number of timbral parameters. Fred Frith's use of the Boss FV-50L pedals for his master volume controls in his performance setup influenced my choice of equipment, and his workshop⁵, which I attended in 2010, reinforced the importance of full control over electronic signal amplitude. I have adapted the use of this pedal to enable it to attenuate any control voltage (CV) signal sent from the synthesizer such as an envelope output or more commonly, a fixed voltage of 5 V so that I can have foot control over any patchable parameter. Such a device also affords volume control over any element in the setup including shortwave radio.

5.3.3 Stepped Filters

Stockhausen's own performance practice for this and related pieces; *Kurzwellen*, *Pole* and *Expo*, featured use of the Maihak W49 filter by the sound projectionist. Although most often used in conjunction with tam-tam and viola, this device is seen in close proximity to the sound projectionist in many photographs from this period. In order to test the hypothesis set out in Chapters 3 and 4 that there are clear parallels of practice between King Tubby and Stockhausen's use of stepped passive filters I incorporated into my setup an Altec 9067b filter set, comprising a 9069b High-pass filter and a 9068b Low-pass filter in series (see Figure 5.1 right hand side). The Altec has rotary controls rather than sliders, making it impossible to control both cut-off frequencies with one hand, but the transitions between frequency steps are characterized by a similar click or crunch as those of the Maihak as heard in the *Mikrophonie I* recordings and as described by Stockhausen (Stockhausen and Kohl, 1996).⁶

To be consistent with the historical practice the sound projectionist should be in control of the filter. Sound projection was invariably Stockhausen's own role in

⁵Conducted with University of Edinburgh postgraduate students in May 2010.

⁶See Figure 4.8 for a comparison of frequency steps.

performances and the filter was therefore his instrument. That makes its associated practice much harder to interrogate since it is only the *Mikrophonie I* score that contains any substantial clues for others relating to the live performance technique. Further study of *Kontakte* may yield other results, but it must be assumed that performance practice evolved considerably even between 1958 and 1966, and carried on until the late 1970s.

5.3.4 Reverb

One of my favourite tools is a spring reverb module. I have customised this to allow the connection of an external spring tank so that I can change the quality of reverberation by interchanging tanks, but mainly so that I can directly manipulate the springs to create electroacoustic effects and extend the range of available timbres. This direct intervention fuses Tubby's dropping and striking of his reverb tank with Stockhausen's extreme manipulation of the tam-tam and Boje's abuse of his own spring reverb, but the nature of the device and the way I have set it up has allowed me to develop my own practice using it.

5.3.5 Sound Projection

Much of Stockhausen's fixed media electronic work uses four-channel, quadraphonic playback for sound projection. The four-channel format is the simplest multi-channel format to achieve practically as it needs at its most basic, only two pairs of speakers.⁷ Part of my interest in presenting *Spiral* in four channels was so that I could design and build a quadraphonic mixer using two Penny & Giles quad panpots that I had found on ebay.⁸ Essentially the mixer comprises two monophonic input channels that can be freely panned using two joystick controllers, and four-channel post pan reverb can be added and combined with the output whose amplitude tracks extremely accurately through the use of VCAs. Designing and building this quadraphonic mixer was a major project and is treated as a portfolio work in its own right, documented in Appendix A.

The important principle was to allow the sound projectionist good control over two monophonic signals and to be able to pan these freely around a three dimensional space without looking at a screen for feedback, thus encouraging visual communication between sound projectionist and performer. Although the first performance was to

⁷Ideally four pairs of speakers are used and configured so as to ensure cover for the whole audience, with the speakers mounted at least 3.5 metres high.

⁸Many of my designs are provoked by opportunistic discoveries such as this, and this is something that I consciously incorporate into my design practice.

take place at the *Soundings* festival in a reverberant concert hall with a twentytwo loudspeaker diffusion system, the four-channel system was the primary sound projection source with other speakers used at the peripheries, for distance, low frequencies and height. Extra reverb was incorporated by using an EMT 140 stereo plate. This model was used by Stockhausen in the realisation of *Kontakte* and so felt like it firmly belonged in the same sound world. The use of tweeter trees suspended above the audience was also reminiscent of King Tubby's technique of suspending the treble horns of his sound system above the crowd at dances.

5.4 Noise and Fidelity

The score provides an interesting perspective on how to approach the idea of noise in performing this piece. Shortwave radio is selected not simply because shortwave radios were ubiquitous in the late 1960s, but particularly because of its noisy qualities. The transformations that radio signals must undergo, first to be transmitted, and then to be received, were of great interest to Stockhausen throughout this period, and the transformation processes of modulation and demodulation are extensively explored in *Telemusik* and *Hymnen* especially, by use of ring modulation and filtering. Boje and Eötvös's recordings of *Spiral* and *Pole* both feature ring modulation as a part of the instrumentation used for performance, and this creates a noisy transformation of whatever material is fed through it.

The score advises the performer to sometimes cut the radio off at the end of an event, and sometimes to allow it to sound at a low volume through several events, providing a background for the instrumental performance. Even whilst searching for a suitable event, the performer must listen quietly, but not silently (i.e. not by using headphones), so that the radio contributes as much to a background noise floor as to active musical events.

The quadraphonic mixer was designed to be as low noise as possible and the design used the highest quality parts available. Any noise entering the signal chain from this point on would have been down to each amplifier or active speaker, the EMT 140 or the ambient sounds in the hall itself. The hum and hiss from the EMT whilst not deliberately exacerbated, was a welcome addition in that it also provided a backdrop of a unique quality so that sounds with long decays could trail off into a generalised noise floor somewhat higher than normal for an electroacoustic concert. As outlined in Chapter 2.2.3, this strategy reclaims some control over the *listening situation* and provides the proscenium arch within which the action may take place.

For the *Soundings* performance I set up the instruments in such a way as to be unable

to attenuate the clicks from the Altec 9067b as I wanted these clicks to be a part of the sound world, although this configuration has subsequently been revised.

5.5 Performance Practice

5.5.1 Configuration



Figure 5.1: Spiral rehearsal setup

In Figure 5.1 the synth is visible in the middle of the picture, the Realistic DX-300 shortwave receiver on the right and the Altec 9067b filter set is the green unit in front of the receiver. Three pedals are visible on the ground, and the ribbon controller is just visible above my arm in the centre of the picture. This basic setup remains the core but certain devices have been added to this such as the external reverb tank and a submixer for better control over levels.

Any performance of *Spiral* must be configured around the ability to receive a reasonable

quality of shortwave signals. For the *Soundings* concert I was able to set up the sixteen metre aerial inside the concert hall and still receive enough of a signal. In ideal circumstances an aerial should be erected on the roof of the building, or at least outside in an open area, however, these conditions can rarely be met in practice, so a compromise is always necessary. I ensured that all computers onstage be shutdown for the duration of my performance and the coaxial cable leading from the receiver to the aerial was another useful measure to prevent RFI from my own electronic equipment.

The requirement to be able to adjust tuning and volume - to perform with the radio as a musical instrument - poses another technical challenge to any performer using more than the voice only, and a mechanical rather than electronic one. Assuming a reliance on the use of the hands by the performer for playing the primary musical instrument, the obvious means of controlling the radio is by the use of footpedals. Tuning controls are often mechanically integrated to the radio's design and are not readily customisable via the use of a remote potentiometer for example. Mechanical extension of control requires advanced fabrication skills and although this would be a useful and rewarding path, my own skills lie more in the electronics field, so my approach was to concentrate on enabling foot control over the volume of the radio whilst freeing up one hand by extending the scope of foot control over more synthesis parameters. This free hand could then be used to control the radio tuning frequency directly.

This meant adding an additional Boss FV-50L volume pedal to control the volume of the radio signal, and adding a repurposed A/B foot-switch, with a 7V signal running through it, attached via an OR gate to the gate input of the synthesizer. This allowed me to use the latching foot-switch to gate the sound of the synthesizer as an alternative to using one hand on the ribbon controller. In turn, this meant that I could adjust frequency or timbral properties of the synthesized sound with one hand whilst simultaneously adjusting the radio frequency with the other. Because the foot-switch is latching, I was also able to use one foot for volume control of the radio and the other for volume or timbral control of the synthesizer simultaneously. This technical arrangement allowed me the potential for a level of control over my instruments commensurate with the requirements of the score, but has also been absorbed into my general practice.

The synthesis material was very versatile and was enhanced by the use of an egg-slicer as suggested by another of Stockhausen's assistants, Hugh Davies (Davies, 2002, pp.52-3). This was amplified using a contact microphone, and a gate signal derived from the pre-amp triggered an ADSR to control output gain via a VCA, and was particularly useful for metallic sounds immediately controllable by touch. This metallic percussive sound can be heard on Portfolio CD track 7, at around 12:20. Two other sound paths

were used and connected via a mixer module on the synth before being passed through the spring reverb module and then to the output.

5.5.2 Technique

A key problem in trying to realise an authentic 1960s version of *Spiral* is the inevitability of picking up music made after that period and speech which clearly is not going to be contemporaneous with that period. Nothing can be done to avoid this so it must be embraced. On a more subtle level, the ambient sound of traffic, aeroplanes and other phenomena audible within the concert hall will have changed considerably in the last forty years, so the suggestion that the sound world is authentic 1970s is clearly untrue. My own practice is dependent to a certain degree on music which I have heard and made so some techniques which I consciously and unconsciously employ may have their origins post 1970.

One event type that encouraged a development of technique was the echo indication “E.” The score instructs that “**An Echo** of the previous is event is to be played/sung” but that the parameters of duration, dynamics and number of segments are “comparatively free” (Stockhausen, 1973). This allows the interpreter to think of an echo in terms, perhaps, of a fragmented signal bouncing off distant cliffs, or a peal of thunder bouncing off a mountain’s ridges and, gradually descending into subsonic rumblings as it gets channelled through the valleys, the air in between absorbing and attenuating the high frequencies. My strategy for these events was to increase the amount of spring reverb signal with added feedback in the synthesizer, then use the Altec 9067b to filter out much of the high frequency content of the signal to create the illusion of distance⁹, coordinating with the sound projectionist via a visual cue to feed the signal to the EMT 140 plate reverb, and to use a combination of the distant speakers in the hall.

Stockhausen advised the use of reverb with electronic instruments in seminars on his Intuitive Music,¹⁰ and Boje and Eötvös both use what sounds very much like spring reverb in their recordings of this piece.

Use of the filter can contribute significantly to the enhancement of Register changes which in the score refer to higher or lower pitch transpositions. Given the likelihood of unpitched sound events emanating from the radio and being made possible to emulate by the incorporation of noise generators, distortion, modulation etc. in the synthesizer, it is useful to be able to think of a pitch-centre in Schaefferian terms (Schaeffer, 2002),

⁹By setting the 9068b low-pass filter at 500 Hz cut-off frequency and switching the 9069b high-pass filter to the “Off” position.

¹⁰Related in seminars at the Stockhausen summer courses in Kürten, 2010 by Hans Tutschku and Intuitive Music Weimar.

and therefore the use of high and low-pass filters can change the perceived pitch centre of an enharmonic or complex event quite efficiently. For sequential “+” signs relating to Register I would increase the high-pass filter frequency by one or two steps each time while playing a noise-based sound with the appropriate amplitude envelope.

The clicks and crunches that can occur between filter steps and which are an inseparable characteristic of the stepped filter must be accepted if the frequency is to be changed dynamically, and I had to incorporate these into my performance. During performance there were occasions when I wanted to adjust the filter in the silent space between events and the clicks felt wrong in these instances. Since talking to Rolf Gehlhaar (see Chapter 4.4.2) I have modified this technique to align my practice more closely historically with that of the Stockhausen Ensemble by placing the filter before the volume control pedal. These clicks can be heard particularly at 5:50 and 15:18 on Portfolio CD track 7.

5.6 Analysis and Summary

5.6.1 Analysis of *Soundings* Performance

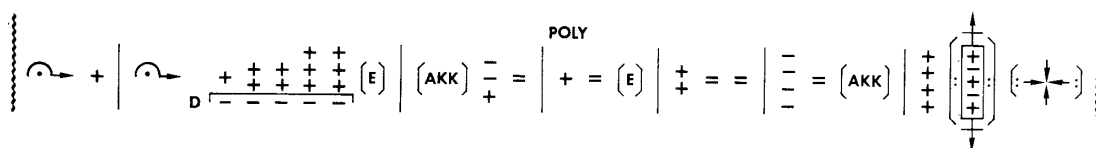


Figure 5.2: Section of *Spiral* selected for performance. (Stockhausen, 1973)

An integral version of *Spiral* can easily last over an hour so to keep it to around fifteen minutes I selected the section above as my version. Having made a few notes for assignment of certain parameters I allowed myself the freedom to assign others in realtime. Deciding which parameters to transform in realtime was very difficult, especially given the repercussions of these decisions due to the highly structural nature of the score. For example, events 4-8 each require a decreased Duration, which means that the first event needs to be fairly long or the duration will become too short too quickly. Preparing the score by attributing specific parameters to each event seems like a step away from the improvised ideal, and Gehlhaar did not prepare to this extent, however, like Vetter, I have found it to be necessary for my own practice in order to be able to retain a fidelity to the score. The big problem is that a slavish adherence to the score can result in as low fidelity an experience as a too liberal interpretation. Stephens and Vetter both advise that whilst the score must be followed clearly, the interpretation must be musical.

I managed to find some code broadcast, a lot of noise, some music and speech,

and using single-sideband-modulation I got some characteristically modulated sounds as well. The problem of searching musically, and then deciding at what point an event has started and then to work out how to emulate it, to emulate it and then to remember clearly the structure and content of that event whilst quickly deciding which parameters to transform in such a way as not to lead yourself down a blind alley is enormously challenging and merits far more rehearsal time. It also leads to incredibly valuable realisations about the nature of listening and responding to musical events in improvisatory, or as Stockhausen might have put it, “intuitive” musical situations.

Both during the performance itself and listening to the recording, it sounds like the most successful shortwave events are those in which the volume of the shortwave sound is adjusted deliberately as in the shaping of an envelope which is suggested by Stockhausen in the examples contained in the score. Synchronicity can be achieved with the output volume envelope of the synthesizer fairly easily since both sounds are operated by foot control. This way, emulation of the shortwave event can be allowed to diverge from the melodic structure of the event but can still retain amplitude, envelope and durational similarities, thus aiding the monophonic instrument to emulate polyphonic sound events.

Despite the preparation, the unpredictability of the shortwave sources made it difficult to play in front of an audience. My own assessment of the performance was that it only really started sounding like a piece of music in the last third. A colleague remarked that the pauses were all the same length and all too long, and on listening to the recording this becomes evident very early on. The result is that the piece remains too disjointed, and with the paucity of sonic structure in the first few shortwave events combined with the “=” points in the score which direct the player to play an event the same as the preceding one, this led to the failure of the first half of the performance on aesthetic grounds. It underlines the importance of maximising the opportunities to receive a good quality shortwave signal, but it also highlights a significant technology led influence on the way the piece sounds forty years after its composition. In some of these cases it might have been prudent to take a more elastic approach to the piece by ensuring that more suitable events were created in response even to a fairly anodyne shortwave event, or even by using several pre-tuned shortwave radios.

Another feature that did not work well in performance was the low level shortwave sound that can continue throughout the spaces between events. This sound plays the role of the canvas upon which other sounds are layered. It also allows longer pauses by the performer to be sustained within a musical context, and it claims the background noise as a deliberate part of the composition. In a similar way to that used by Portishead and described by Frith (Frith, 1996), the background noise can become most effective

when it is removed entirely. Similar to the low level of music playback suggested by Eno for listening to his *Discreet Music* (Brian Eno, 1975) (see Chapter 2.2.1), this was at a level which was good onstage but which was not going to the front of house and so was far too subtle to be heard properly by the audience. In the concert hall this noise was so quiet as to become masked by the general room tone. This can easily be remedied in future performances.

This problem led to more interrogation of the effectiveness of the FV-50L volume pedals. I have found that while they work well for imparting expressive dynamic amplitude control over signals, it is very tricky to use them to control overall volume level accurately as well as with the same feel that an acoustic instrumentalist might have. For this reason I have started to incorporate a secondary knob or fader as a final master gain control so that changes in Intensity are more easily realised, whilst the pedal can continue be used through its full range for dynamic envelope control and expression. The relationship between fader curves is significant in that it is clear through practice that a volume pedal has characteristics which can only be appreciated fully through performance. I may yet take the soldering iron to these pedals...

5.6.2 Recording Analysis

My practice has evolved out of the use of repurposed, used, fixed and broken equipment and has expanded to include and embrace rather than reject or subdue environmental factors. Having spent some time configuring my stage setup to accommodate separate outputs for front-of-house and for recording, and setting up a stereo microphone in a Blumlein pattern, the recording was supposed to provide the potential for a comprehensive mixing session to fine tune various parts of the performance at a later date. Unfortunately my assistant failed to enable the direct recording tracks so only the live microphones were recorded to the computer. I had, however, setup the Studer A80 to record the two discreet line outputs from the synthesizer and the radio to tape, so the direct sound was captured to a medium again contemporaneous with the composition.

Synchronizing these takes was difficult and required numerous edits as the tape speed varied enough to create a flanging sound and pre-delay when heard against the microphone audio.¹¹ To continue the experiment of exploring Stockhausen's techniques I used the quad panner to simulate use of the rotation table for panning a four channel mix, using the Deutsche Grammophone recording of *Kontakte* (Stockhausen, 1968) as a model. This recording was made by Stockhausen for stereo from the four channel tapes and features some sounds which are heard outside the lateral position of the

¹¹I really need a new pinch roller and bearings to fix this.

loudspeakers on a good sound system. Examining the photographs of Stockhausen at work using the rotation table it is evident that for sound directed at any given microphone - e.g. front left, a 180° phase inversion of the same signal, attenuated and with reduced high frequency content, will be picked up by the opposite microphone; in this case the rear right. If rear left and right signals are mixed to the front for a stereo mix, this would provide one explanation of the audible effect of such reverse phased signals as heard on the record. Stockhausen used phase changes that were produced as a result of this early rotation table as well as the later much larger machine built and used for *Sirius*, although these were exclusively acoustic phenomena. (Stockhausen and Maconie, 1989, p.148), (Stockhausen and Kohl, 1996, p.90)

For my stereo mix I concentrated on the individual source channels and routed each through one pannable channel of the quad panner. The front left and right outputs were panned about 80% left and right in the mix and the rear channels were hard panned, phase inverted to the **opposite** sides, about 12 dB down with an additional 8 dB attenuation of high frequencies using a shelving filter set to 4 kHz. This allowed for normal stereo panning with the joystick in positions between full front-left and front-right, with the additional ability to stretch the perceived stereo field beyond the width of the loudspeakers when pulling the joystick back slightly and engaging the rear left and right outputs due to a combination of phase inversion and an approximated head transfer function. All of this was under direct tactile and latency-free control, using absolute positioning of the joysticks, allowing screen-free and immediate performance within what sounded in the studio like a surround environment. The results heard on the portfolio CD (track 7) showcase various panning techniques. Without being able to rely on many hours of practice using these techniques, it is inevitable that some are more successful than others, however, they do achieve a measure of virtual surround perspective reminiscent of the Deutsche Grammophon recording of *Kontakte* (Karlheinz Stockhausen, 1960) and provide a foundation upon which I can build up future practice.

For added effect I needed to use the microphone tracks but as the hall was large and the diffusion system naturally contributes various delay anomalies as part of its inherent acoustic characteristics, the constantly shifting time variance between the direct and ambient sounds just added to the spaciousness of the recording. The EMT 140 was employed in the mix to accentuate the echo events in the score even more, and the associated noise present from having the EMT returns permanently engaged, along with the albeit tiny amounts of tape hiss contributed a useful background to partly mitigate the absence of continued radio sounds through some of the gaps between events. Some print-through can be heard at various sections on the recording but I accept this as a welcome addition to the background noise, making up, as it does, for

the lack of low-level shortwave sounds. Although not deliberately adding noise at the *production environment* level, I was happy to allow it into the recording and mixing, embodying the various material characteristics of some of the different instruments, devices and bodies that went into the production of the piece.

5.6.3 Summary

An understanding of shortwave, aeralis and RFI is vital to increasing the chances of picking up useful shortwave signals. I have seen a performance of *Kurzwellen* with each performer using a laptop right next to a shortwave radio, with the inevitable result being a horrible soup of almost formless noise and static, revealing very few signals from which to derive musical events. One such rehearsal performance was even scheuled for 10 am, a time at which it is virtually impossible to receive shortwave signals due to the Sun's interaction with the ionosphere. This was not helped by the use of ring modulator models on the laptops to further distort the shortwave sound, as if the shortwave sounds themselves were somehow lacking in complexity and subtlety.

Immediate and intuitive control of both the shortwave radio signals and the instrument is vital, and even a little technical knowledge about shortwave can give the performer a much greater chance of finding suitable material to transform. This is vital because it is the sonic foundation of any interpretation, but perhaps the most valuable lesson learned from performing *Spiral* relates to listening.

The most difficult challenge of the piece is the combination of skills needed to operate the radio whilst emulating or augmenting the events generated, remembering these with enough detail to be able to emulate them and transform them, and doing all of this in a recognisable but musical fashion. Technical design and control, whilst vital ingredients, are useless without considerable performance skill, and the requirement to transform events in a musical way is as close to composition as it is possible to be. Events may suggest themselves from the shortwave, but working out, in realtime, which parts of which events to focus on, combine, and use, is reliant on a compositional skill beyond that of a traditional score interpreter. The skills of a free improviser are also inadequate by themselves to play *Spiral* since the score must be followed and set structural relationships must be created accordingly.

Technical practice must be at a sufficient level to allow fast emulation of and response to events, but such practice is really stretched by this piece, especially if the Spiral-signs are followed literally, "... TRANSCEND [the event] ... BEYOND THE LIMITS OF YOUR INSTRUMENT." (Stockhausen, 1973) But also suggested by the score is the use of "bellows, 'Waltzen', tape loops, electronic storage systems, acoustical and optical electronic controls, time-delay apparatus" (Stockhausen, 1973) all of which

would require even greater extension of technique, but some of which could be used to great effect. In a similar way to *Mikrophonie I* Stockhausen leaves the door open for technical development to influence interpretation, but in the less determinate score for *Spiral* there is perhaps much greater scope for this to be done within the bounds of authenticity.

Some results cannot easily be communicated if they relate to unmeasurable (“perceived” (Clarke, 2005)) phenomena such as, in this case, an influence on the performer’s listening practice with respect to judgement about performance decisions using the filter steps, or judgement about the likelihood of noisy or quiet transitions between steps and their suitability at particular points in the performance. During the performance I was aware of the clicks as the frequency was changed whilst playing the Spiral-sign event and felt that this sounded appropriate. However, when switching filter frequencies during pauses, the clicks were an unwelcome addition to the music which, in retrospect and after talking to Rolf Gehlhaar, influenced me to change my setup. Rather than trying to eliminate such noises from the recording, in response to Chapter 2.4.5, I feel that these sounds should remain in the recordings as evidence of physical performance and interaction with this electronic instrument. The adaptation I have made to my listening practice both as a performer and a recording engineer/producer is to accept these sounds. Cleaning these up in post-production would sterilize the experience and remove its connections to the wider sonic environment. Minimising them at opportune moments in the manner described by Gehlhaar is an option based firmly on existing practice formed through a vast amount of playing experience and that therefore seems a reasonable technique to employ.

One sound that caused more concern during the performance and which was also audible in the recordings was the sound of my foot being removed from the volume pedal (CD track 7, 0:50, 2:32, 3:40...). I like to have bare feet while playing since this increases the tactility of my contact with the pedals, however, in this case my foot kept sticking to one of the pedals and made a noise when removed. Since this is not an inherent characteristic of the pedal I feel that this should be avoided in the future, although twenty years from now this sound could be subject to exactly this kind of study of the traces of physical performance, and I might well feel different about it then. This is also the case with the sounds at 4:56 and 11:18 where I turn the receiver on with a click after having accidentally turned it all the way off previously. These are the sorts of sounds I enjoy discovering in other recordings as described in Chapter 2.4.5, and they are part of the sounds of these repurposed devices used as musical instruments.

The tacit knowledge that arises through such experimentation is the very information which is lacking in the written accounts and the scores, but with the benefit of experience gained through performing *Spiral* I have not only learned lessons about this

piece but have also been able to identify the right questions to ask of past performers of the piece in order to tease out more details of related practices.

One of the most unusual features in the score is the instruction relating to the Spiral-sign:

FROM THIS POINT RETAIN WHAT YOU HAVE EXPERIENCED
IN THE EXTENSION OF YOUR LIMITS, AND USE IT IN THIS AND
ALL FUTURE PERFORMANCES OF “**SPIRAL**.”

(Stockhausen, 1973, p.14)

My own experience from having played *Spiral* is that in the last section, playing the Spiral-sign event and the following contracting/diminishing event, I felt that I had learned a powerful lesson about how to listen to my own and others' playing and how to incorporate structural as well as timbral elements within my response to this. I felt as though this section of the performance was at last musical rather than procedural, and that I was really exploiting my instrument to something near its true capabilities, in large part thanks to the modifications, reconfigurations and new modules built in response to the challenge of performing *Spiral*. I found myself actively exploring the transitional area between low frequency pitch and high frequency rhythm within a defined musical context of the development and transformation of a specific phrase, bringing to bear some of Stockhausen's fundamental ideas about fixed electronic music (Eimert and Stockhausen, 1958) in realtime by using technology contemporaneous with the composition of the piece. I consciously absorbed the noises from radio, the machines, and my own body and made them part of my practice, and was able to successfully incorporate King Tubby's filter into an authentic performance practice of Stockhausen's music.

The Spiral-sign also makes unequivocal the idea that practice is constantly evolving. It expresses as a command the necessity to build on existing practice and extend it, as opposed to just using different ideas all the time, and therefore resonates very strongly with the main motivating factor behind my thesis. *Spiral* will be a part of my repertoire for many years because of this demand to keep extending my limits, and its impact is bound to be felt throughout the rest of my work.

Chapter 6

Portfolio and Conclusion

6.1 Other Portfolio Works

Whilst the course of this research seems to have been directed towards a musicological outcome, my own practice has been maintained as a fundamental component. Many details, both coarse and fine, would have remained obscured without the constant incorporation of my findings and testing of my hypotheses through my extensive improvisatory practice and through the remainder of the pieces documented below. Rather than presenting an album or some such polished body of work, some of these pieces I consider to be able to stand as successful compositions, installations, or artefacts, and some are to be considered for their value as documented experiments.

Throughout my practice I have always sought to engage with an ethical approach by making use of discarded or broken items, literally and metaphorically. I have incorporated into my designs parts of a broken electric golf trolley, wood reclaimed from a discarded Victorian oak wardrobe, electronic components from an old burglar alarm, re-purposed RGB monitor cables... In my performances I have used record players rescued from skips, records from charity shops, and many other machines which I have had to repair before use.

This level of technical involvement has been necessary both for my performances using old technology to take place, and to enable me to learn about the implementation and influence of all kinds of components and materials on the musical outcome. I have invested time, effort, and resources in building a physical relationship with my materials. This approach is also indicative of an active resistance towards the increasingly consumer led marketing onslaught typified by software updates, sample libraries, generic synthesis plug-ins and cheap MIDI controllers destined to break after

only mild usage. My creative practice enmeshes my composition and performance skills with my technical and design skills, with each element influencing the others. This is expressed in this brief summary of works included in the portfolio, with music documented in this chapter and designs in Appendix A and B. Although the designs are equally as important as my own music and interpretations, their documentation is included in the appendices because of the more technical nature of the material.

Noise features heavily in most of my work, and where I use less noisy material it is usually best described by reference to frequency as opposed to a pitch. Scores are not provided as the pieces are either structured improvisations, fixed media or sound installations. The problem of communicating such works to other performers and therefore contributing to a repertoire is not confronted, the role of the portfolio being much more biased towards understanding existing and historical practice.

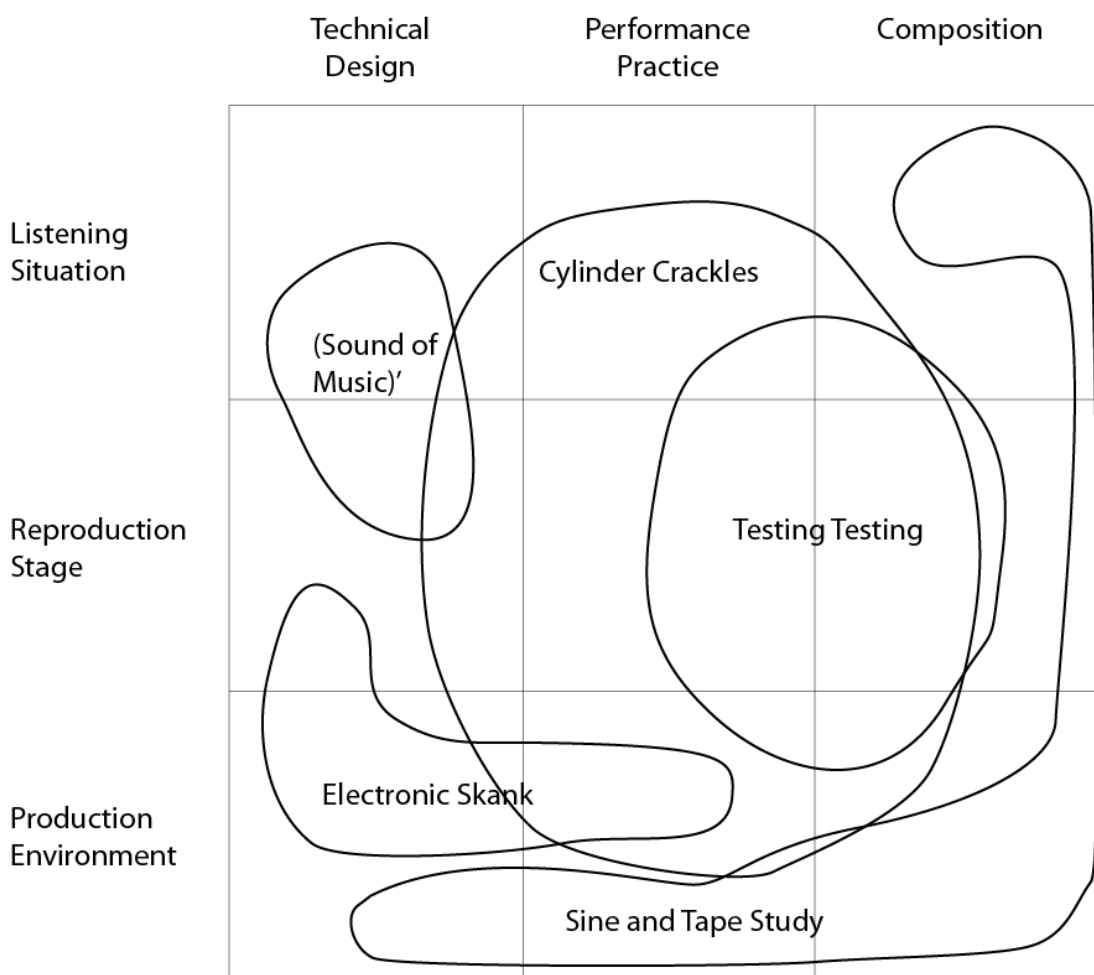


Figure 6.1: Portfolio works in relation to the main ideas explored in the thesis.

I explore all of the areas in which the three archaeological levels of the *listening situation*, the *reproduction stage*, and the *production environment* (see Chapter 2)

intersect along the continuum between technical design, performance practice, and composition as set out in Figure 6.1. The two performed works by Stockhausen and Ono are illustrated in Figure 6.2.

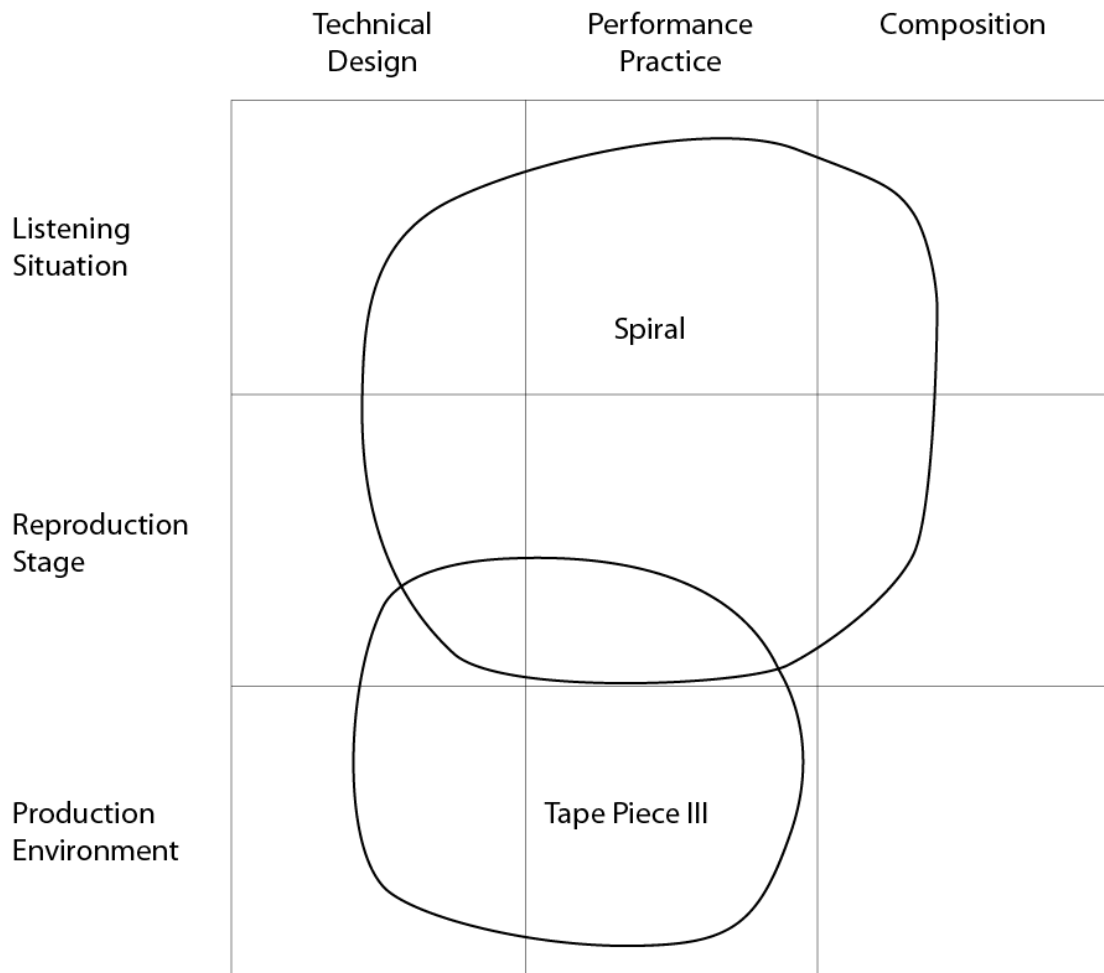


Figure 6.2: Portfolio interpretations in relation to the main ideas explored in the thesis.

6.1.1 Cylinder Pieces - Three Way Conversation, Ghost Tracks

Portfolio CD track 1 and 2
 Portfolio DVD *Cylinder/Cylinder Session Photos.pdf*
 Portfolio DVD *Cylinder/sbkw_open_door/* containing:

- *sbkw_door_end.aif* - Short end sound file
- *sbkw_door_main.aif* - Main sound file
- *sbkw_opendoor_demo.maxpat* - Demonstration patch showing simulation
- *sbkw_opendoor_patch.maxpat* - Master patch to be used for installation

As I have tried to express in Figure 6.1, all of the ideas examined in my research contributed to the realisation of this piece. Starting from an impressionistic design response to the idea behind the STEIM Cracklesynth (designed by Michel Waisvisz, 1975), and partly inspired by Bowers' infra-instruments (Bowers and Archer, 2005) I created a circuit that was incredibly difficult to use in a predictable way. The design and manufacture of a small electronic sound making instrument based on linking up circuitry from three STEIM Cracklebox-type circuits is the foundation of the instrumental composition of this piece. It is an attempt deliberately to extend the composition into the technical design realm, and vice versa, by producing an instrument so unpredictable that it is a continuing negotiation to play this instrument and repetition of precise sounds is almost if not totally impossible. This precludes its use to interpret existing pieces and also makes it difficult to use even with other improvising musicians, although I have used it in performance with harp and bass clarinet.¹



Figure 6.3: Cylinder recording session. (Image courtesy of Aleks Kolkowski)

In May 2011 I was invited by Aleks Kolkowski to contribute two pieces to his Phonographies archive project (Kolkowski, 2011). I used this instrument to record a two minute piece acoustically onto an Edison Cylinder, making full use of the inherent surface noise of the recording medium to form an integral part of the composition.

¹A performance of Edimprov in the Reid Hall in May 2010.

Constrained by the medium to around two minutes, and uncertain as to how the instrument would behave, we recorded one cylinder as the first piece, and I tried to coax the instrument into converging with the cylinder's own sounds as well as trying to create some sustained tones to help draw attention to the speed fluctuations of the mechanical recording and playback process.

The second piece was prompted by Kolkowski's experimentation with a sound-on-sound technique:

[We] superimposed two individual recordings on the same cylinder using a proto-overdubbing technique that inscribes two spiral grooves, one over the other. The stylus 'reads' both grooves simultaneously, usually with frequent echoes, intermittent distortion and skipping effects as the grooves merge into each other. Here, however, the result is remarkably even.

(Kolkowski, 2011)

These pieces only really exist as one-off wax cylinders and can therefore only be authentically experienced by arranging a visit to Aleks Kolkowski and having him play them back acoustically. This is still somewhat easier and certainly cheaper than arranging for an orchestra to perform a symphony, and the included digital recordings, which are simply for documentation purposes, can be considered as the equivalent of a Sibelius type rendition of a traditional score using a sample library instead of a performance with real instruments. Here, the medium of the recording is one of the three elements comprising the identity of the piece, alongside the electronic instrument and the negotiation enacted by the human performer.

The digital copy has been used in an installation for Julijonas Urbonas' Open Door project at ZKM in Karlsruhe where a main door at ZKM and another at the Hauptbahnhof were fitted with a sensor and a loudspeaker (Urbonas, 2011). I made a Max patch that plays back the file from a point determined by the velocity with which the door is opened, which gives the impression of a great variety of sounds, different for each door-user. When the door shuts, passing a certain threshold, the sound is crossfaded into a second recording of the needle coming to the end of a wax cylinder noisily. My `sbkw_opendoor_patch.maxpat` was the patch used for installation and my `sbkw_opendoor_demo.maxpat` patch demonstrates this patch using Urbonas' door simulation sub-patch. The choice of Max/MSP is not particularly essential to this piece but it does allow easy installation.

To hear these pieces in high fidelity the reader may contact Aleks Kolkowski via the Phonographies website² and arrange an appointment directly.

²<http://www.phonographies.org/contact/>

6.1.2 Sine and Tape Study

Portfolio CD track 3

Sine and Tape Study takes as its materials the technique of creating transient noise by tape splicing at 90° and the listening situation of the theoretically perfect sine waves after mediation through reproduction medium of tape and acoustic space. Rather than writing a piece ideally executed according to the algorithm, I have used algorithmic techniques (LISP/Common Music) to expose and amplify the characteristics imposed on real, magnetically transduced and acoustically presented sine waves by these ecological effects and the effects of technique. This piece is a pre-cursor to a more ambitious future study of Stockhausen's *Studie II* (Stockhausen, 1956) which inspired the techniques I used to create this piece. The process consisted of four main stages:

- Stage 1
Record twelve different sine waves onto lengths of tape, the frequency of each being a fibonacci series number multiplied by 100 (units in Hz): 100, 100, 200, 300, 500, 800, 1.3 k, 2.1 k, 3.4 k, 5.5 k, 8.9 k, 14.4 k.
- Stage 2
Cut sections of these tapes to any one of twelve different lengths from 1" to 12" using aleatoric methods and splice these with twelve different lengths of spaces created using leader tape. The distribution of lengths of tapes should be biased towards shorter lengths. (See *sine_tape_sketches.pdf* on Portfolio DVD).
- Stage 3
Playback the resulting tape in a concert hall accompanied by an improvised flute part using an array of five microphones to maximise the unusual characteristics of the hall (St Celilia's Hall, Edinburgh). (Listen to *tape_solo_recording.aif* and *tape_flute_recording.aif* on Portfolio DVD for the two recorded versions.)
- Stage 4
Using a simple sample-rate playback variation instrument in LISP/Common Music, decode the recordings using a specially prepared algorithm that transposes the recording at the start of each new sine tone so that each sine tone is transposed to the same frequency (500 Hz) with durations lasting until the start of the next new sine tone but modified by the change in playback rate. Reverberation causes the sound to overlap and this creates polyphonic material, and the flute improvisation is subjected to the same transpositions thus completely changing its structure, range and timbre, in many cases, beyond recognition. (See Appendix D for the decoding algorithm).

In setting up the tape machine to playback sounds in the concert hall I had to make decisions about speaker and microphone placement that forced me to confront noise in various guises. St Cecilia's Hall has a glass domed roof and seagulls in particular as well as traffic, sirens and other external sounds are often present. The shape of the hall means that several nodes exist whereby if a speaker is situated in an opposite node to a microphone (or listener) a very pronounced and focused flutter echo is created. The tape machine, although quiet, makes some mechanical noise in operation, and I placed a microphone near the tape machine in order to have the option at the mixing stage of including the feint sonic traces of the machine's physical rather than its purely electronic presence.

An unexpected result of combining the physical activity of tape splicing and note taking in order to assemble the stimulation tape and using LISP to process the results was the breakdown in accuracy of the analogue process made apparent by the decoding algorithm. The decoding algorithm relies on the measurements of sine tape and leader tape as notated during the long editing process, to transpose the recording made of the sine tape being played in a space to only one fundamental frequency. This means that if the chosen master frequency is 500 Hz, any portions of tape-plus-space at 100 Hz would be played back five times faster, and any portions at 89 KHz would be played back 17.8 times slower. Ideally the encoding and decoding process should yield a series of perfect 500 Hz sine waves, but the subject of the piece is the amount of timbral difference exhibited by executing the process physically. Tape edits, wow and flutter, recording pre-amps, room acoustics, microphone selection and placement contribute to a "worldization" (Ondaatje and Murch, 2004) of the process.

Because of mistakes in the logging of measurements and inaccuracies in the measurements building up over the length of the tape, the decoding algorithm started to produce results which deviated from the master frequency centre. This departure from the plan sounded more interesting than the subsequent version produced after the logging errors had been identified so a compromise was made and an intermediate version used because it sounded better. A further deviation from the strict plan was the inclusion of Richard Worth's flute playing, recorded as a guided but improvised addition to the sine wave tape in homage to Ussachevsky and Loenig's early tape experiments. The extreme transpositions of the recorded flute add the required human element and variability to the sonic material and transform the piece from a potentially dry technical demonstration or experiment into a working piece of music.

This piece acts as a prism through which to explore an instrument sounding in space, and should be a good foundation for a whole series of work. The main disadvantage to my approach was in using LISP/Common Music as I have not been able to get it working again for over a year. The software is beset with bugs and installation requires

the patience of a sage. Since I lost a hard drive in 2010 and my G4 laptop died in 2011 I have been unable to install LISP/Common Music on my current machine so have been unable to develop this further.³

Included on the Portfolio DVD are the compositional working sketches *Sine Tape Study/Sine Sketches.pdf* as well as a recording of the sine tape in the space and another of the sine tape plus flute before being “decoded” by the LISP algorithm.

6.1.3 Testing, Testing

Portfolio CD track 4

Portfolio DVD *Testing Testing/Testing Excerpt.mov*

The video is an excerpt of a performance in Edinburgh and the audio recording is a studio version.

Testing Testing takes similar material - test tone recordings - and through repurposing similar in ideology to Tubby and Stockhausen, uses these outside their testing/laboratory context creating a musical context in which their imperfections are welcomed, emphasized, and exploited. This piece, again, is about the traces of human physical interaction with the tools of production, both as evidenced in the repurposed tools themselves (cf. the clicks of the stepped filters) and in the live performance - the needle being dragged sideways across the grooves.

Using test tone records, test oscillators and a homemade ring modulator, this piece explores the musical use not only of test equipment but of test media. These otherwise sterile sources are artefacts that have been designed to be as perfect as possible, but this piece revolves around the inherent imperfections due to design, ageing and physical presence. Sine waves on 60 year old records are no longer mathematically pure, and the ring modulation shows up more of these defects by amplifying differences in an interesting way. The ring modulators (see Appendix B.3), based on the simplest (and cheapest) of designs using two centre-tapped transformers and four germanium diodes, are also not perfect theoretical multipliers. Indeed, this type of design, favoured by Hugh Davies (Davies, 1976), is sometimes used for its more noisy behaviour. The physicality of the record stylus is also brought to the fore in this piece, again using the ring modulator as an audio lens to magnify the characteristics which lie on the boundaries of practice. Very low frequency vibrations of the stylus create a tremelo effect when fed to the ring modulator (due to amplitude modulation), and this is exploited in the performance.

³I should point out that the tape part still works fine.

This piece was developed in parallel with practice by finding interesting areas of interaction between various test tone records and the ring modulation processes. A test tone tape was used in performance for added drama using a remote control operated Studer A62⁴, and reverb was also found to be useful, especially when using a footpedal to control reverb input gain. A structure for the piece was worked out and successfully adhered to but subsequent attempts to produce a more detailed performance score were less successful since this was based more on a transcription of one performance and was felt to be unable to communicate the required feeling to an interpreter. The piece relies more on liberal interpretation and a sensitivity towards the material and how to transform it for an audience, and as such, this remains difficult to pass on to other performers.

Almost any test tone records can be used for the piece as long as at least one has a long descending swept tone. The sine waves are of primary importance but test records with noise, warble tones and samples of music are also required. The voices of the narrators on such records lend an air of authority to the piece, and recited frequency notes give an impression of a countdown-to-lift-off. The key record that should be used is *How to Give Yourself a Stereo Check-Out* (Decca, 1967) narrated by Jack De Manio and Elizabeth Knight. This has a great range of tones and signals but also has some very funny dialogue. The piece ends with a small portion from *Derek and Clive, Live* (Derek and Clive, 1976).

The signal flow is set out in Figure 6.4, with signals routed to the ring modulators via aux sends on the mixing desk.

There is one detail of the technological performance practice that has no bearing whatsoever on how the piece sounds or how the audience reacts relates to the test oscillators. I use Levell TG150 oscillators for this piece. These are the same as the ones used at Abbey Road Studios in the 1960s to control the varispeed on one of the tape machines used for ADT, flanging, and phasing during Beatles recording sessions. John, Paul, George and Ringo and many other engineers, tape operators and others would have gently wiggled the perspex arm of the control knob of this oscillator during bounces and mixdowns to generate these effects (Ryan and Kehew, 2006), and I get great enjoyment from emulating this historical physical performance practice during performances of this piece even though the audible effect is totally different. This is visible at 2:45 in the video excerpt on the Portfolio DVD.

Appendix B.3 contains design information about the Ring Modulators built in conjunction with this piece.

⁴One of the first solid state tape machines, launched by Studer in 1964.

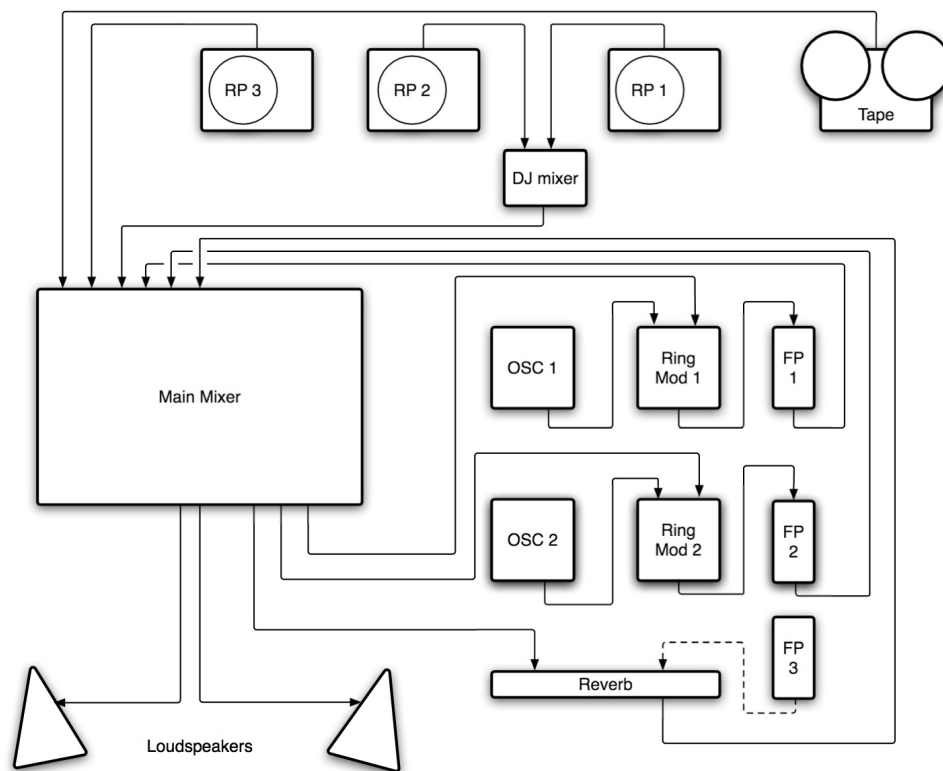


Figure 6.4: Testing Testing signal flow

6.1.4 (Sound of Music)'

Portfolio DVD *Sound of Music/Sound of Music documentation.mov* and *Sound of Music/jit.op-xfade_sbkw.maxpat*

The effect of new technologies on listening practice has been approached through an interrogation of noise; noise being very often the signifier of the boundary conditions which represent the edges or borders of such practices and technologies. *(Sound of Music)'* interrogates the way we listen to vinyl - how reproduction media act on our perception of music and what the medium contributes to what we are listening to. It holds a lens to the extra musical phenomena contributed by the medium as a part of the listening ecosystem. This is the blank canvas of Lucio Fontana or the Movie Theatre photographs of Hiroshi Sugimoto but rendered in sound.

This is an installation piece based on a forensic examination of the surface noise of a vinyl record - a mono copy of *The Sound of Music* (Rodgers And Hammerstein, 1965). By inverting the phase of one side of the signal from a stereo cartridge and mixing it together with the signal from the other side it is possible to cancel out almost all of the original music, only leaving the difference between the two signals, i.e. the scratches, scuffs and imperfections, the remnants of previous owners' interactions with

the artefact. This sound is then played back over a sound system in a darkened room with the signal triggering a Max/MSP/Jitter patch to display occasional flash frames of a scene from the film.

The cancellation process is not perfect with every record so in this installation I have used a Drawmer DS-301 noise gate to help suppress the sound of the music below a certain threshold. Experiments with other records, in particular a 7" single of Lee Perry's mix of Junior Byles' *Beat Down Babylon* (Junior Byles, 1984) (track 12 on the Examples CD) have yielded better cancellation which leads me to question the quality of the cutting process of these mono versions of *The Sound of Music* (tracks 13 and 14 on the Examples CD). These three examples all start with a straight mono signal from the left channel of the record player, and then the inverted right channel signal is faded up until the best cancellation occurs. This signal is then faded out towards the end of each example and the remaining signal is then faded out as well.

The title is derived from Set notation, meaning that this is the set of everything except *The Sound of Music*, implying also that the piece itself is not music. The choice of record is based on its ubiquity - the record sold in enormous numbers and enjoyed huge popularity. It can be found in almost every charity shop in Britain and is therefore extremely accessible. In terms of copyright this is also a challenging piece in that the phase reversal process technically removes all of the encoded music on the disk and in theory it is only the distortion, noise and imperfections that are rendered audible by this process.

Documented video was taken at an installation of this work in Dundee, but it is not particularly effective to present a longer recording of the piece since it is designed as an installation rather than a musical work. Copyright issues surround this piece (and a series of other work using customised CDs not presented here), and the full visual element used in the installation is not able to be included for this reason. The Max/MSP/Jitter patch is included but is not fully documented as its role is very simple and any other solution could easily be used instead. All the patch does is to act like a noise gate but for a visual element instead. A video clip is played at a speed selected by the curator, but the projector screen remains dark unless an audio signal from the record player triggers the gate to open, at which point the picture becomes briefly visible.

The piece may be installed using any other mono record, with or without a visual element.

6.1.5 Electronic Skank

Portfolio CD track 5 and 6

Portfolio DVD *Electronic Skank/ES London/Electronic Skank London.mov* and *Electronic Skank/ES Edinburgh/Electronic Skank Edinburgh.mov*

This piece explores King Tubby's tools and techniques by using them in the context of a live improvisation piece for four players. The performers each take on the role of one of the tape channels in a dub mix, drums/percussion, bass/low frequencies, rhythm (guitar, organ, horns), voice, and I perform a live mix of all these elements using the "Big Knob" filter, tape delay and the Fisher spring reverb unit configured as in Tubby's studio. Whilst the final sound of the piece is likely to be very different each time because of the autonomous decisions made by the improvisors, the style imparted by the use of Tubby's techniques gives it its identity. In these two live recordings Lauren Hayes performs "rhythm," Jules Rawlinson "bass," and Owen Green "voice."

Two recordings and accompanying videos of live performances are included here, one recorded in London without access to tape delay but using a modulating digital delay instead, and only three other improvisors instead of four (Portfolio CD track 5), and one recorded in Edinburgh, including in place of a fourth player a tape recording of some UPIC experiments discovered in a box of old 1/4" tapes, made by Peter Nelson in the 1980s at CeMaMu, Paris (Portfolio CD track 6). The material acted in the place of the nominal "rhythm" (guitar, organ and horns) track, and its use as a fixed media element was therefore more authentic in terms of Tubby's practice of mixing from four tracks of fixed material.

The key discovery resulting from this piece was the effectiveness of the combination of delay level and delay feedback controls in one fader (see Chapter 3.3.5). By performing the piece both with and without access to tape delay the particular sonic qualities of the tape delay - its compression of loud signals and the way feedback of the delayed signal is controlled by the tape medium - were able to be appreciated within the performance practice context.

Appendix B.1 contains information about the "Big Knob" filter designed in conjunction with this piece and appendix B.2 contains design information about the Varispeed Remote module also built for this piece.

6.1.6 Yoko Ono - Tape Piece III

This thesis was submitted tied with a length of 1/4" tape as a realisation of *Tape Piece III - Snow Piece*, 1963 from *Grapefruit* (Ono, 1971). The piece is all about

the contemplation of the practice of realisation. In this way it resonates with much of Richard Long's work, displaying traces of evidence of the practice rather than presenting any sounding music at all. It can also be thought of as an extension of practice for my own purposes, or practice for its own sake, as demonstrated in the use of the Levell TG150 for *Testing Testing* (see above: Chapter 6.1.3).

This text piece consists of the following lines:

Take a tape of the sound of the snow falling.
 This should be done in the evening.
 Do not listen to the tape.
 Cut it and use it as strings to tie gifts with.

(Ono, 1971)

It is vital for my own listening experience that I feel some kind of connection with an element of physical performance as an implicit (or explicit) part of the music I hear. I value the traces of interaction of a human being in what I can hear. This final portfolio piece addresses only the idea of this relationship between a person and technological means of making music. It deals exclusively with the culture/technology alignment of analogue sound recording and is engaged only with the process, technique, and execution. It took over two years to realise this piece and the journey to its completion was characterised with false starts, false promises, and maddening failures of computer technology.⁵ The pace of progress meant that I lived with the piece for a long time and in order to realise this version I had to find and fix a more portable tape machine (a discarded Revox A77) which I could keep at home so that I could be in a position to exploit the opportunity when it arose.

6.2 Conclusion

My archaeological approach has provided some useful strategies for analysing the communication of all kinds of information regarding the performance practice and other conditions relating to music production through noise. Much of the territory relating the three archaeological levels - the *listening situation*, the *reproduction stage*, and the *production environment* (see Chapter 2) - to the three elements of the electronic music-making continuum - composition, performance practice and technological design - is creatively explored within the portfolio (see Figures 6.1 and 6.2).

⁵In the first year I had the Studer A80 and microphones ready to go but all of the equipment got stuck in the lift overnight after the University card-reading system failed. Another opportunity did not arise for twelve months.

The principal practical outcome of this research has been a documentation of the sonic potential of the Maihak W49 filter and the associated performance practice relating to its use in *Mikrophonie I*. Much still needs to be done in order to understand how it was used in performances of other pieces, but I have been able to offer some foundations for any such further study, partly by building a replica interface and partly by incorporating a similar stepped filter into my own interpretation of *Spiral*. I have shown how characteristics particular to the W49/W66c combination have influenced the performance practice of *Mikrophonie I* and by extension, further pieces, but at the same time I have drawn attention to the existence of a class of such devices, albeit a very small class.

I have identified the instruments used by King Tubby and by first recreating and subsequently using similar equipment, I have tested and explored ways in which these instruments were configured in the studio in order to make records sound the way they did. The observation of very close parallel practices, as exemplified in King Tubby's use of the Altec 9069b filter, supports my argument that there is considerable influence exerted over the musical outcome, including through interactions with performance practice and composition, by such technology. Rather than being determined by the technology, I have argued that this influence is exerted principally through the alignment with the technology of a set of craft, design and performance skills and abilities.

By identifying and examining to the component level the Maihak and Altec filters I have been able to demonstrate how they interact in synergy with the other performance elements used in each context. I have considered primarily measurable elements such as impedance, signal gain-makeup, physical resistance to movement, and have made observations on construction design, component choice, and other technical details in as comprehensive a way as possible. Some data has led to a re-evaluation of the probable choices behind the use of these instruments by the music makers, particularly; Stockhausen's changing attitude towards the noise and clicks inherent in the repurposed use of the W49, and his subsequent incorporation of this into his practice; King Tubby's use of what was available rather than the more mythologised ideal of his having made all of his instruments himself.

Manning observes that certain electroacoustic pieces by very different composers can still be identified as being produced in particular studios because of the use of certain items of equipment (Manning, 1999), but I wonder whether this fully acknowledges the influences of the studio technicians and the practice. Could it be that what is recognisable is the evidence of a culture/technology alignment between technicians, composers and instruments, within the context of studio performance practices? Many studios have an Altec 9069b but none sound like Tubby's studio. Few studios have

W49 filters but nobody produced music sounding like *Mikrophonie I*. Born's accounts of the influence of the IRCAM tutors over their composers' work is worth taking into account when considering such influence in other situations (Born, 1995).

This supports my application of Hill's concept of the *alignment* between knowledge, whether tacit or theoretical, and tools as the critical factor determining the practice and musical output, and not simply the availability or affordances of the tools or instruments themselves. The affordances of the Altec 9069b in the MCI mixing desk did not change between its installation in Byron Lee's Dynamic Sounds studio and King Tubby's Dromilly Avenue studio yet the difference in the nature of the musical output from the two studios is enormous. It was the alignment of Tubby's technical knowledge, improvisatory technical and musical practice and exploratory nature with the potential of the Altec filter and his other devices that led to the repurposing of them all and their coalescing into a unique musical instrument. His practice, once developed, was then robust enough to be passed down to Prince Jammy, Scientist, Philip Smart and others and to maintain its recognisable form even in the hands of others.

Tubby moved from technician to performer and, by 1981, back to technician, (Taylor, 2008) having innovated a new practice and handed it down to his apprentices. In the light of his untimely death in 1989 we can only speculate as to the influence this performance practice had on Tubby's technical and design practice. Stockhausen, however, exhibits a different movement along the continuum between technician and composer. He employs the skills of technicians and assistants (who often remain unnamed and uncredited) to develop his compositional and in *Mikrophonie I*, his performance ideas. After becoming directly technically involved in studio realisation, tape editing, and the mechanics of electronic music production in the 1950s, he later retreats to a more directorial compositional role, and becomes less technically involved with the production of sounds. We may trace the influence of his performance practice in his compositions, perhaps expressed by the use of deliberate noise elements in his later formula based compositions comprising the *Licht* cycle (Stockhausen, 1981). Part of the super-formula consists of the technique of unvoiced or noise based sounds to be played on the trumpet, trombone, basset horn, flute and other instruments. We may associate this in part with the noisier elements of the W49 filter which he used in the studio and in sound projection for over twenty years. In these later works, the technique of stepped filtering is also prominent, achieved acoustically through the use by horn players of a variety of different mutes.⁶

As tacit technical knowledge is so important to both these practices, it is significant that the assignment of credit is markedly different but equally problematic in both domains. The engineers and technicians behind Stockhausen's practice and realisations are often

⁶worn on a holster around the waist for easy access.

very difficult to identify since their skills are often not documented and their names not always recorded. There is a distinct hierarchy with composer at the top, performers in the middle, and technicians at the bottom. Werner Scholz's contribution to the sound of the Stockhausen Ensemble for example, resulting from his choice of modules and design of the pre-amp rack that was used throughout the late 1960s and 1970s and therefore central to the aesthetic of the sound quality, is significant and yet unknown to all but the most persistent researcher. Gehlhaar's self-disparaging comment⁷, his reluctance to ascribe value to his own contribution to the design of the W49 filter instrument by fixing the two W66c faders to it with two brass strips, is an indication perhaps of a prevailing attitude towards such contributions by the creators and designers themselves, and perhaps helps to explain why such information is so hard to find.

King Tubby is often credited on record labels, sometimes in the title, sometimes as producer, sometimes as engineer, yet there is a good deal of confusion and debate about whether certain mixes were done by Tubby, Jammy, Scientist, or others. With much of the studio output being generically credited to Tubby, this, in effect, ascribes more priority to Tubby's technical expertise than to the individual mix engineer/performer. Writing credits for individual tracks are ascribed to songwriters or producers⁸ and rarely include any credit for Tubby's creative input in copyright terms. In both cases there is obviously an issue of branding which influences the scrupulously fair attribution of credit, but especially in this style of music with creative input distributed over a wide range of places, people and influences, there is still no satisfactory working model of creativity that can be applied successfully to the commercial implementation of credit and copyright (Morey and McIntyre, 2011). In the light of the obvious clumsiness and unfairness of the common models, especially the composer, performer, technician model, this makes it all the more important to find out who was responsible for such influential contributions to the sonic outcomes of this practice.

By finding out the material details of electronic instruments, as presented in Chapters 3 and 4, I have demonstrated that it is possible to trace the influence of these devices in the musical results of their use, and to redress the balance by tracing these contributions back to the engineers and technicians responsible for their design and implementation. To identify the composer as the sole creative contributor to music making and to ignore the technical expertise and practice that, by its alignment with the technology enables such music to be realised, is to miss the possibility of understanding this music as a collaborative process.

Building and incorporating these kinds of devices into my own practice has allowed me

⁷ "Big deal!" (Gehlhaar and Williams, 2011) (see Chapter 4.3.1)

⁸ This area in Jamaican music is especially resistant to untangling due to the control exercised by producers, fixers, and other vested interests - not much different to other music industry practices, but perhaps more extreme.

not only to test hypotheses and extend my own practice, but has also been vital to enabling me to ask Gehlhaar and Müller the right questions. Without this foundation in practice, I would have missed the significance of such details as the cleaning of the faders (see Chapter 4.5.1) or the combination of delay level and feedback controls (see Chapters 3.3.5 and 6.1.5).

Without rebuilding the W49 interface (see Appendix G), without particular attention to the two-faders-in-one-slot design, and without finding the W66c faders, valuable insights into both the setup and performance practice associated with *Mikrophonie I* and other pieces would not have been possible to glean. One of the most satisfying moments in this research was seeing Stockhausen's W49 instruments at the WDR Studio, and bearing in mind I had only seen black and white photographs before this, seeing and touching the bright brass strips holding the W66c faders to the main body of the unit, and being able to connect this directly with Gehlhaar's account of how he fixed them together with brass strips himself. "Big deal" indeed! The other was encountering King Tubby's mixing desk at the Experience Music Project in Seattle and seeing for myself the heavy wear patterns made by Tubby's hands around the filter control and channel faders - the evidence of how he used the instrument.

Confirming that both Stockhausen's and Tubby's filters use capacitors and inductors to create their sonic results is fundamentally important in supporting my argument that these electronic devices are material musical instruments. An inductor is a piece of wire wrapped around a magnetic core. A capacitor is made from two thin metal plates separated by an insulating material. These are both simple material components, depending for their electrical characteristics on the nature of the materials from which they are made, and the way in which they are arranged and put together. This, along with the "carbon strips" to which Stockhausen refers, and the other physical components of these instruments is what has a direct material effect on the oscillations of the electrical signals passing through them, which we hear as music, and this is something possible to understand at the tacit level of the practice of making electronic music. Tubby built amplifiers and sound systems; he made records; he wound his own transformers and therefore had a creative and physical experience of this level of technology. His creative studio practice as evidenced by his many records links both these areas of craft in one individual.

Even as early as the 1970s we find Lancaster referring to active filter circuits (using op-amps) as "models" of inductor/capacitor circuits (Lancaster, 1975), decades before accessible computer modelling techniques which have all but replaced the use of physical instruments. Arguments in favour of modelling tend to employ the accusation of nostalgia against practice based on analogue instruments, and without wishing to start a debate at this stage, whilst nostalgia is a factor, both in technical practice, and

listening practice (as demonstrated in Chapter 2.4.3) there are many more processes at work which are often overlooked perhaps because of the strength of the prevailing market led alignment between contemporary digital practice and technology.

Being able to think materially about material goods, hence critically,
gives one some independence from the manipulations of marketing...
Knowing the production narrative, or at least being able to plausibly
imagine it, renders the social narrative of the advertisement less potent.

(Crawford, 2009, pp. 17-18)

Crawford here implies a social narrative within the practice, and I engage with this in my own practice by weaving in my own narratives such as the use of the Levell oscillators (see Chapter 6.1.3), by not editing out the accidental noises of my feet on the pedals (see Chapter 5.6.3), or by using as a subject the history of the handling of a piece of vinyl by its previous owners (see Chapter 6.1.4). I observe this narrative in the practice of others both as a listener (see Chapter 2.4.5) and as a researcher. Hearing the evidence of materials in music reminds me that people made this in a social interaction often involving many other people, and by combining as much evidence as possible regarding the materials and techniques I can enhance my appreciation and enjoyment of the creative processes involved. This extends beyond the composer and the performer to the technician as well - all those involved in the craft of electronic music making. Understanding these processes enables my own practice to evolve accordingly and gives me the opportunity to transmit what I have learned as I am trying to do in this thesis.

For me, the most interesting material is the sonic evidence of music makers' physical interaction with the electronic sound making equipment, and the noisy traces that transmit this physical relationship beyond the *production environment* through the *reproduction media* and the *listening situation* into the ear, brain and consciousness of the listener, brilliantly illustrated and made visible by the wear patterns on Tubby's MCI desk. The imperfections resulting from the struggle to make machines serve a purpose exceeding their original design limitations is a direct indication of the presence of human will and action in the creative music making process. This creative, molding connection, extending all the way back past the performer to the design, engineering and building of the instruments themselves, in this case by the staff at Maihak GMBH, and Grover C. "Jeep" Harned at MCI, and probably Art Davis at Cinema Engineering and Altec, is a vital part of what constitutes this music.

In analysing the instruments to this level of detail I have moved my own practice along the path from being a user/consumer of music technology equipment to being a fixer/maker of electronic musical instruments, and my creative output has been

transformed in the process. In the course of this research I have come to regard electronic music as a craft. This understanding down to a component level has been achieved in parallel with significant changes in my listening practice, fundamentally connected with the consideration of listening and performance being part of a continuum and not so easily separable. I have come to accept noise, in its many guises, as a necessary component of music making and music listening, and I have been able to engage with it and use it in a number of different ways. The mechanisms by which it can exert influence, as explored in Chapter 2, are a stark reminder of what may be lost as a result of its elimination. The drive of the prevailing culture/technology alignment to achieve greater and greater efficiency, exemplified by the elimination of the human operator as the last obstacle to efficiency in the system (Hill, 1988), the proliferation of software emulations and sample libraries, and in listening circles, by the hi-fi mentality and all the tone tests of the last hundred years, make it all the more significant to offer resistance by properly examining what is being discarded so quickly and so ruthlessly, and by questioning the value of noise.

I believe that progress can be made by exploring the use of new technologies whilst at the same time building on experience already gained through several decades of existing practice. I have demonstrated that understanding the materials, instruments and systems helps us to understand practices in such a way as to be able to build on these and adapt them to our own needs. Carrying out research such as this is therefore essential to the future development of a robust electronic music practice in which creative control and agency are not ceded to hardware or software manufacturers, but remain the domain of the creative practitioner. Future research in this area will attempt to discern and transmit valuable lessons learned through tacit knowledge and experience, and allow future practitioners to build on this well developed historical practice, conserving the momentum of those who have gone before rather than blundering along blissfully unaware.

Despite the level of detail observed by examining the electronic musical instruments in these case studies, much of the information regarding practice would have been impossible to derive without reference, in the first instance, to the video of Prince Jammy performing a mix on the MCI desk, and in the second instance, to Rolf Gehlhaar's demonstration of the performance techniques he used with the W49/W66c instrument. There is thus a great urgency to continue research in this area whilst it is still possible to directly consult primary sources, i.e. the composers, performers, and technicians involved in all facets of the production of this electronic music, before such information is lost forever.

Appendix A

Quad Panner Design

Please see the Portfolio DVD *PhD Materials/Quad Panner.pdf* for working design sketches and more photographs of the unit.

Design and construction of a two input four channel quadraphonic mixer.

- Two fully quad pannable input channels.
- Four discreet hard-panned input channels with adjustable level for fixed media sound. projection or four channel aux return input.
- VCA based master output gain controlled by a single fader.
- Small footprint for minimum intrusion in audience sweet spot.
- Very low parts count for cleanest possible signal flow.
- Low cost
- Highest possible quality physical interface

A.1 Specification

Based around two Penny & Giles quad panning joysticks, I wanted to design a small footprint mixer capable of freely panning two monophonic signals with a post-pan auxilliary send and four discreet input/return channels.

Amongst the many sites of choice in the design process, the selection of four channel output was the most important. In order to setup four channel playback it is only necessary as a minimum requirement to have two pairs of stereo amplifiers and loudspeakers. This is a relatively straightforward technical requirement and so makes

any music written for such a system much easier to have performed than any more complicated setup such as 8 channels or a multi array diffusion setup. The four-channel setup has the additional benefit of being almost universally standardised to situating speakers at each corner of a square, with two in front and two behind the audience.

Immediate, latency-free physical control over the panning, sensitive and accurate control over the overall sound level and the ability to balance the output were all fundamental requirements, and the footprint of the mixer was to be as small as possible, ideally small enough for the sound projectionist to occupy only one seat in the auditorium, thereby allowing more audience members to be closer to the sweet-spot in the middle of the square made by the speakers.

The mixer was designed and built around the chance acquisition of a pair of Penny & Giles (P&G) quad panpot joysticks. These can be used as passive devices but in order to have complete control and the highest possible sound quality I had to build an active mixer. P&G faders from an old Neve console were also sourced cheaply via the internet and for overall volume control, rather than trying to find a four-channel fader I cannibalised some old Studer 990 mixer channels by removing four Studer 1.911.292 VCA cards (see Figure A.1), controlling them all with one linear P&G fader so that the master output volume would track accurately for each channel.¹

Being constrained by the panning law designed into the P&G panpot allowed the rest of the design to progress reasonably quickly without too much time spent experimenting. A critical detail that was overlooked initially was the optimum output load of 47 k Ω required by the quad panpots, and so an additional board containing two quad buffers (one following each panpot) presenting a 47 k Ω load impedance to the panpot had to be added later. The panning law had been adversely affected without the right impedance leading to the image collapsing too readily when panned between speakers. The experience of having to fix this oversight really made tangible the importance of considering the behaviour of such components within a system - i.e. their synergetic behaviour. Just like the Altec 9069b, the quad panpots were sensitive to the output impedance/load with audible results. The turret containing the 47k load resistors and buffer amps is shown in Figure A.2.

In order to achieve the best possible signal quality - i.e. as near to a piece of wire as possible - I had to use as few op-amp stages as possible. This meant an absence of inserts, direct outputs, monitor outputs, EQ, and only as many stages as were necessary to allow the correct polarisation of signals. This must be taken into account throughout the design process as different op-amp configurations lead to different results. When

¹Even with a good quality stereo fader, anything better than 5% accuracy in tracking is unusual.

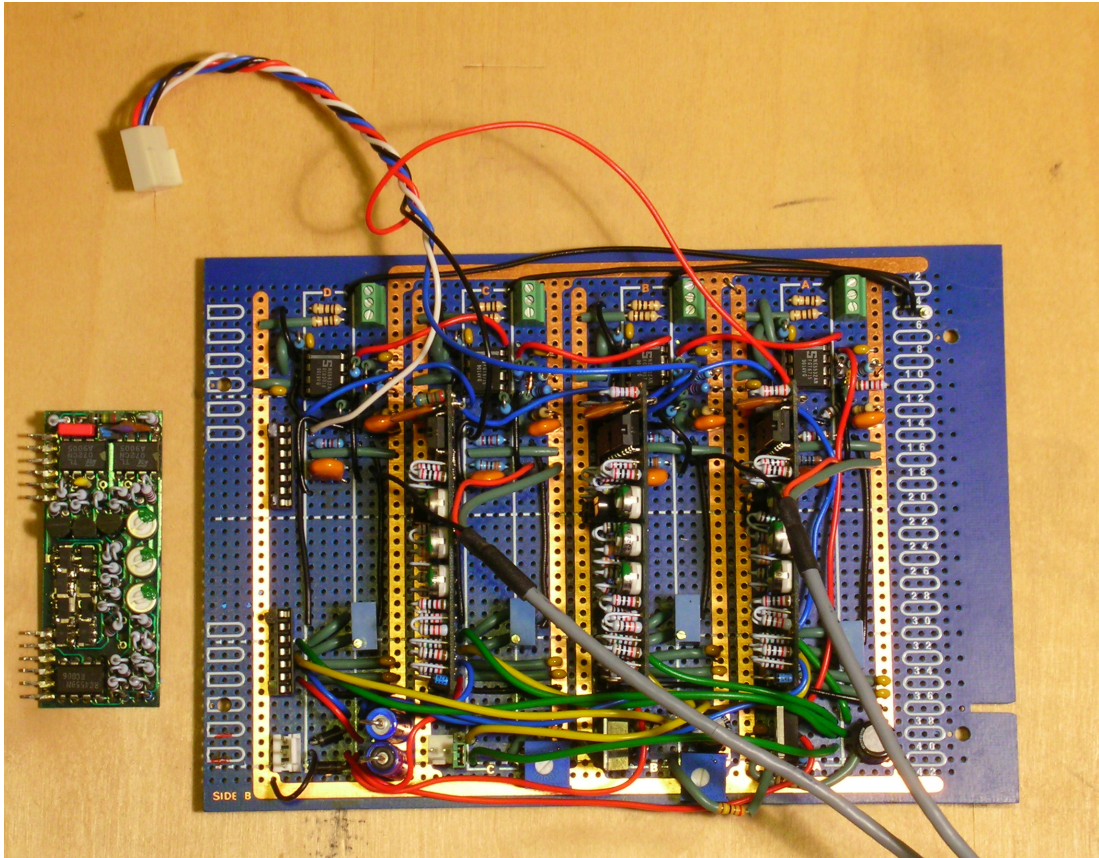


Figure A.1: VCA and output driver Board with Studer 1.911.292 VCA to left.

configured as summing amps, the net output is inverted, a simple buffer amp will be non-inverting, and a differential amplifier (an amplifier that accepts a balanced signal and outputs an unbalanced signal) can easily be made to invert or not simply by swapping the inputs. Another design principle that I tried to adhere to was the general feeling that it is better to use an inverting opamp for gain.

The Neve faders came equipped with a gain stage (using an NE5534) and so that was included in the design rather than building something similar from scratch but this caused an issue as described below. The NE5532 was chosen as the main op-amp because of its excellent heritage and its ability to drive a $600\ \Omega$ load whilst remaining within spec. Most of the NE5532s were also cannibalised from the Studer channels. For the summing amplifiers which did not need to drive a low impedance load I used OP275s as they are also high performance op-amps with very low noise specs, but even these are quoted to be able to drive $600\ \Omega$. In such cases however, the tried and tested choices are better employed since manufacturers have been known to quote specifications that are right at the edge of acceptable limits and not necessarily entirely reliable. The post panpot buffer amps needed to have a small footprint and also did not need to drive a low impedance load. Since these were only buffering and not amplifying I chose

the more generic TL074 - a perfectly decent op-amp without having such high specs as the others, but used in a role in which such high specs would not be necessary.

Figure A.2 shows the additional board containing the panpot buffer amps that had to be added to the fader and master summing amp board. This also contains the connection to the post-pan auxiliary send module, although this picture was taken before the subsequent addition of the coupling capacitors on each channel.

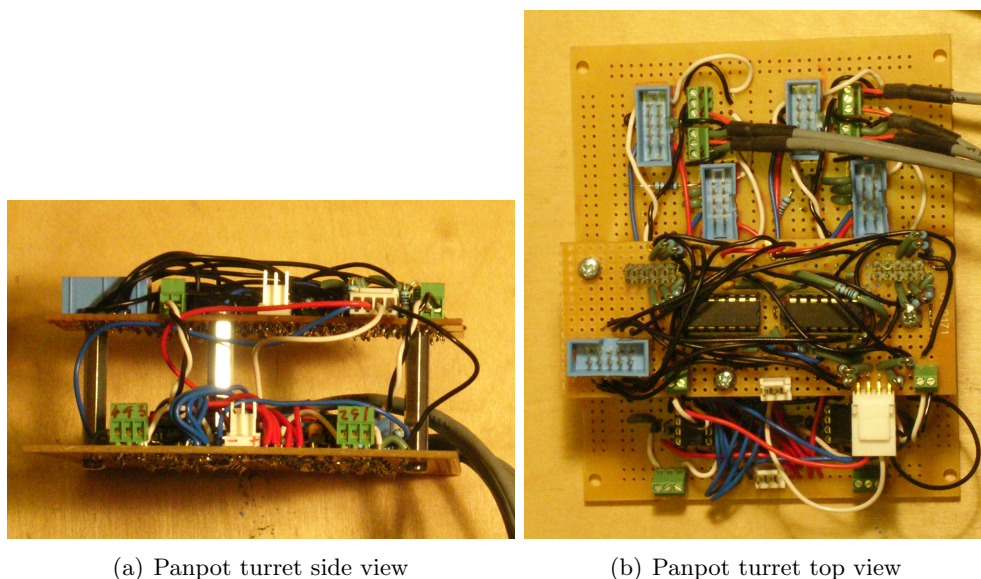


Figure A.2: Panpot and fader connection, output buffer, and master summing amp board

The majority of electrolytic capacitors I used were Nichicon or Panasonic, selected in accordance with commonly held perceptions of superior audio quality amongst online forums, but I tried to eliminate the need for any capacitors in the audio path at all. Unfortunately, on measuring the assembled unit for testing and calibration, I discovered a maximum of almost 200mV of DC offset at the outputs with full gain (+10dB). This was traced to the Neve circuit and so it became necessary to capacitively couple the post-fader signals using two electrolytic capacitors biased from the +15V rail through a 1 M Ω resistor - this was a design borrowed from Graham Hinton. The Neve mixing desks had transformers at most stages so DC offset would have been eliminated in that way, but it was still surprising to find the Neve card being the source of such a significant problem rather than my own designs.

If more space had been available I would have used a DC servo design in order to eliminate DC offset without having to resort to capacitors, however, the problem was only discovered at a late stage in the design process so this was not possible. For similar reasons of space saving the main outputs were designed as unbalanced, with XLR-pin 1 connected to the chassis. Again, this meant one or two fewer op-amp stages and the

configuration still allows for noise rejection in the equipment receiving the signal. Inputs and Outputs are all on two 25-pin D-sub connectors for ease of installation.

Each op-amp's positive and negative power inputs were decoupled to ground using 1 μ F multilayered ceramic capacitors to ensure a minimum of noise wherever it might be likely to creep into the circuit. This means the inclusion of around 40 extra components but they act as insurance against the encroachment of noise where it is not wanted and also have a tiny footprint.

One other component that dictated some elements of the design was the half-Danner size Neumann auxilliary send module. This has two pairs of balanced pre/off/post switches and potentiometers, the signal path in German broadcast equipment being balanced as standard. Since the signal-path in my mixer is unbalanced I was able to convert this into four stereo pairs of controls, thus allowing post-pan auxilliary send for each of the two pannable channels.

A.2 Physical Interface

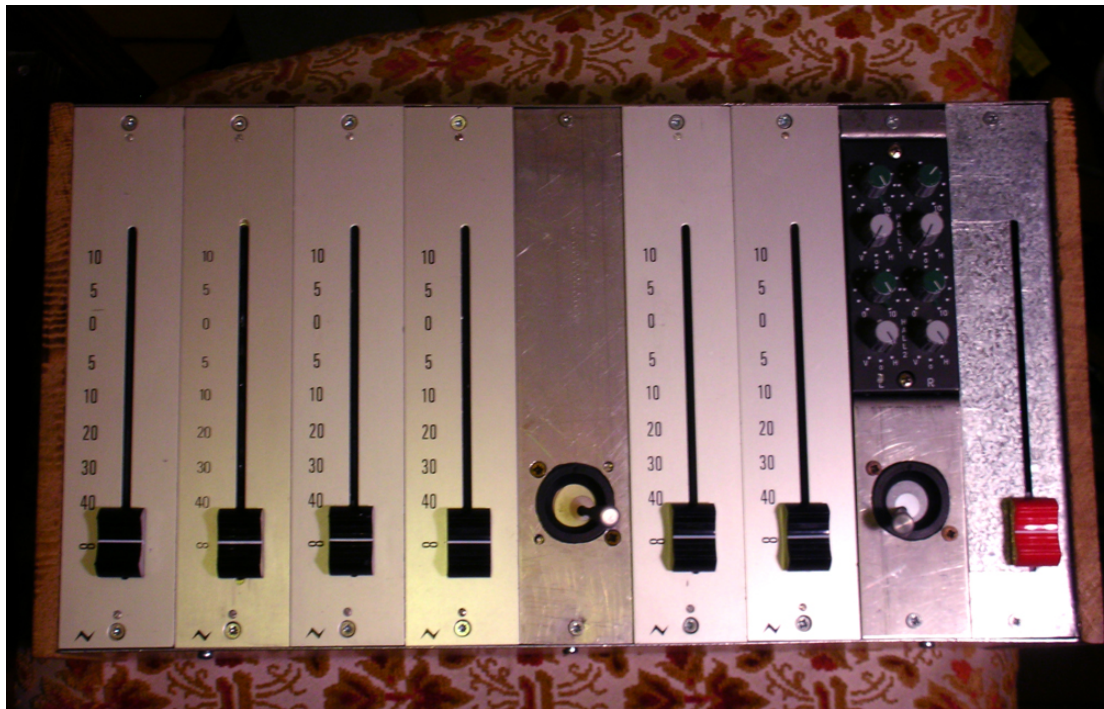


Figure A.3: Quad Mixer control surface

The Mixer is setup in such a way as to accept discreet four-channel line level inputs, providing each with a fader for balancing. Channel 1 goes to front-right output, 2 to front-left, 3 to rear-left, and 4 to rear-right. Two additional input channels feature

fader volume control and post fader quad-panpot with a post-pan auxilliary send. The front and rear L/R signals can be independently routed to Front, off, or Rear aux-send, this allowing a four-channel post-pan signal to feed a stereo reverb unit for example. Normally two stereo reverb units would be used, one for the Front L/R signal and the other for the Rear L/R signal.

The four discreet channels and the two post-pan channels are summed and fed to the four main output VCAs which are controlled via the master fader. Thus two operational modes are envisaged - one simply providing balance for a fixed media four channel piece using only the four fixed channels, and one using the two panning channels with auxilliary send to four-channel effects which are returned to the discreet channels and mixed with the two pannable signals. For more complicated setups the mixer can easily be used in conjunction with a larger mixing desk. In order to provide more options for routing it would have been necessary to make the mixer physically larger and this would have diverged too much from the purpose of the design - i.e. keeping the footprint as small as possible.

A.3 Improvements

Most extra features would demand more controls and would necessitate either building a new housing, or utilising the remaining faceplate of the first quad panpot. An extra port for two more D-sub connectors was made and covered with a blank panel, so some input/output expansion is possible.

There is the possibility of adding external VCA control globally via a jack socket, or individually to each VCA from a modular synthesizer for example. The design includes a summing amp for the CV signal, and the possibility of controlling each VCA independently by the inclusion of jumpers in the original wiring. These jumpers can be removed, and with installation of an additional op-amp in the free space on the circuit board it would be possible to control each VCA independently. If the mixer is to be used in conjunction with my synthesizer I may carry out this modification.

Since there are only six input channels there is also the potential to use the two free channels of the 8-way D-sub input connector for monitor outputs for stage monitoring purposes. This could be done with a couple of concentric pots and another dual op-amp and would make the system more robust.

A useful addition would be some sort of LED or lamp to notify the sound projectionist that the unit is powered up. One of my design principles was to have no distracting illumination or visual feedback, but there have been occasions where this would have been useful.

The mild steel of the housing is unfinished and as such, wherever it has come into contact with skin, has started to rust. Although this cosmetic result looks fine to me, I will have to rub it down with wire wool and spray the unit with black enamel paint to stop this happening. The rust can easily stain clothing and spraying the unit black will also help reduce its visibility when in use. The rust is a visible reminder of the fact that the instrument is not strictly a fixed device, and that its sonic characteristics will inevitably change in time, just like the records used for *(Sound of Music)*² and *Testing Testing* and the two Cylinder recordings in the portfolio.

I built in extra circuit protection using diodes and capacitors for the ± 15 V power supply on the VCA board but because I had no room to do this for the other two powered boards I ended up with a voltage drop on this board compared to the others. This is not easy to remedy and I will only address this on the other boards if absolutely necessary.

A.4 Results

The sound quality of the mixer is excellent² and the unit has been used successfully by me on a number of occasions in performance as well as by Chris Watson and Owen Green. The immediate control over the quad panning is entirely intuitive and the 10 dB of gain available on each channel are just about enough to boost lower level signals. It has also been used for mixing virtual quadraphonic sound into stereo by using phase and EQ on the rear channels (see Chapter 5).

A.5 Improvisations

The layout of the channels and panpots went through several revisions and the design sketches show how the layout evolved slightly (see Portfolio DVD *PhD Materials/Quad Panner.pdf*). The positioning of the two panpots either side of their respective faders allows for a certain amount of control over both volume and panning at the same time, and the eventual layout was found to be the most ergonomic. Part of this was dictated by the height of the circuit boards containing the VCAs which meant that the panning channels had to be on one side only, due to the extra depth of the quad panpots. The symmetrical setup of the panning channels allows for easy two handed operation, although it is still not very easy to manipulate both joystick and channel fader at once with only one hand. A compromise would be to insert a volume pedal on each panning channel before the signal enters the mixer thus allowing more control.

²although I am obviously quite biased.

The wooden end-cheeks were sourced from a scrapped oak wardrobe and are therefore resilient to damage and consistent with the recycling bias of my practice. The belated discovery of the DC offset problem forced me to rewire the summing amp board to accommodate the electrolytic capacitors, and the realisation that the panning laws of the panpots were being affected by the incorrect load meant that I had to construct a new board to mount on top of the summing amp board.

The biggest problem in assembling the unit was the amount of board-to-board wiring that was necessary. I could have reduced the wiring problems by deciding on a multipin connection system early on, yet this might have constrained my choices too much. As with any prototype, there must be room for flexibility in the interpretation and implementation of the design, but if another mixer were to be built this would definitely be addressed. The U-shaped design of the metal housing means that to access the internal components, the faders must be unscrewed and removed. Since both board assemblies consist of two stacked boards each, extra disassembly is necessary to access the lower boards, making testing quite tricky. The VCA circuitry and the ± 10 V supply circuitry is all on the upper board under the main channel faders and so calibration is straightforward. Other boards have no calibration controls.

Space has been left for some future modifications and it is assumed that the nature of such modifications will only be revealed through practice. As such, the design is not fixed and it acknowledges the potential for customisation. The addition of two monitor sends is a priority but the easiest addition that I am contemplating is the ability to add another D-sub output passively multiplied to the main and auxilliary outputs to be connected to an 8-channel ppm for visual monitoring.

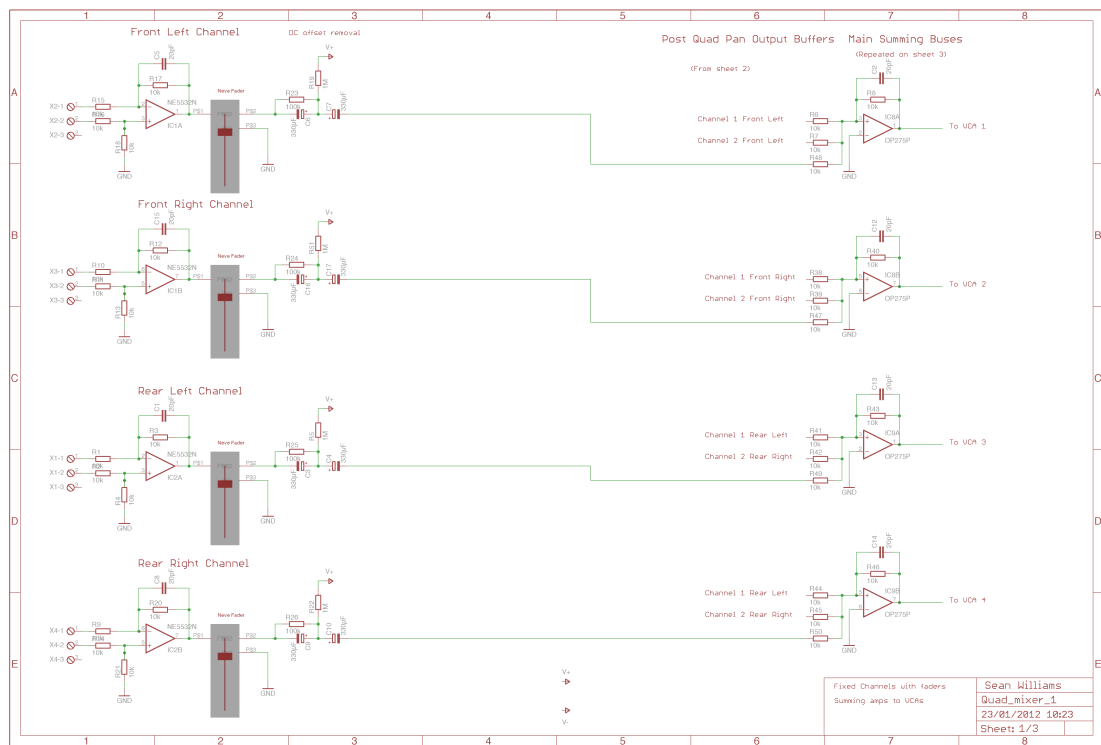


Figure A.4: Quad Panner schematic 1 of 3.

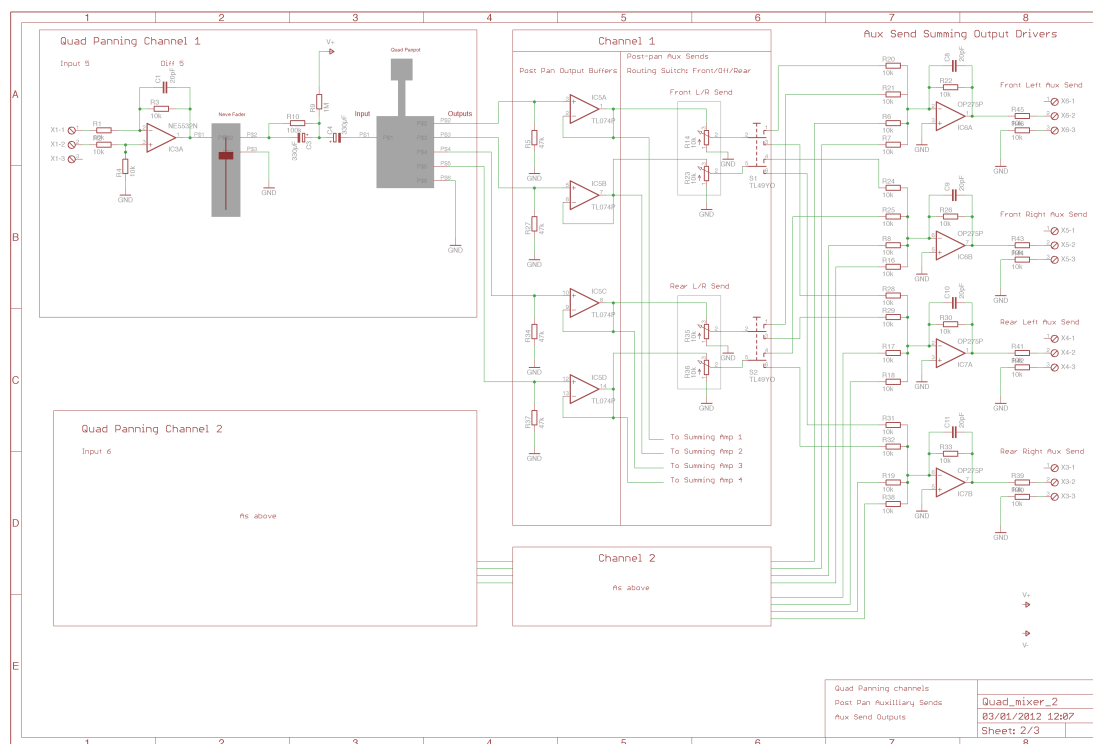


Figure A.5: Quad Panner schematic 2 of 3.

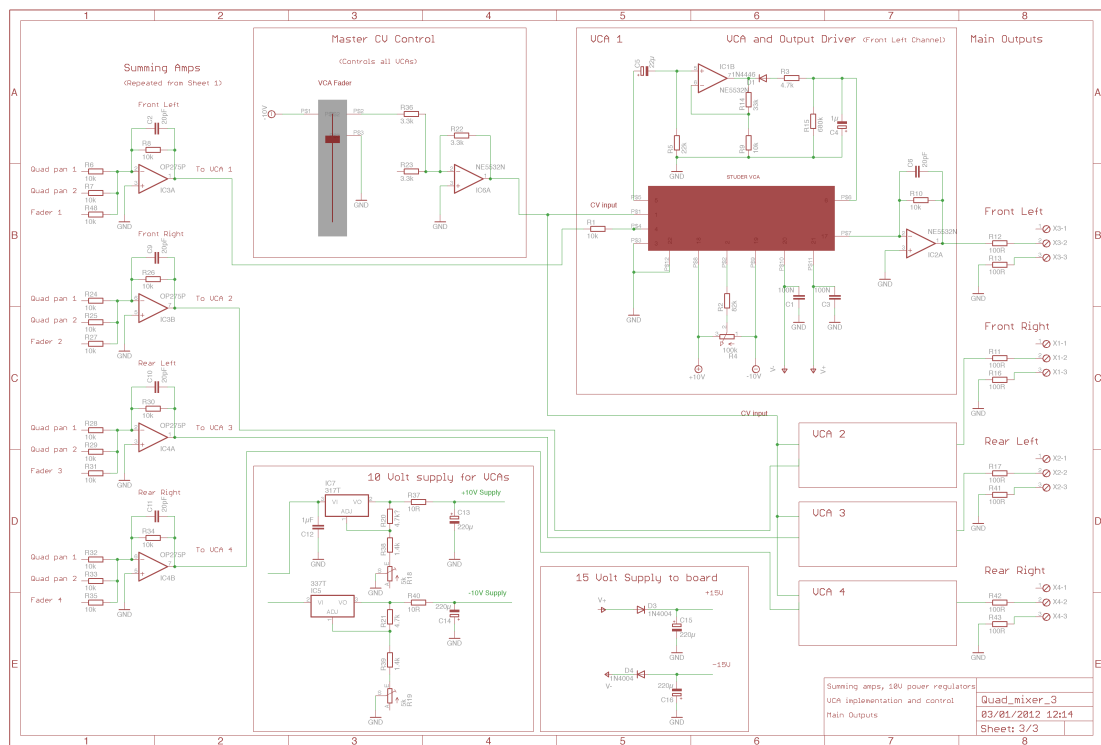


Figure A.6: Quad Panner schematic 3 of 3.

Table A.1: Quad Panner Parts List

Part Number	Value	Sheet	Comment
C1	20pF	1	
C2	20pF	1	
C3	330 μ F	1	electrolytic
C4	330 μ F	1	electrolytic
C5	20pF	1	
C6	330 μ F	1	electrolytic
C7	330 μ F	1	electrolytic
C8	20pF	1	
C9	330 μ F	1	electrolytic
C10	330 μ F	1	electrolytic
C12	20pF	1	
C13	20pF	1	
C14	20pF	1	
C15	20pF	1	
C16	330 μ F	1	electrolytic
C17	330 μ F	1	electrolytic
IC1	NE5532N	1	
IC2	NE5532N	1	
IC8	OP275P	1	
IC9	OP275P	1	
R1	10k	1	
R2	10k	1	
R3	10k	1	
R4	10k	1	
R5	1M	1	
R6	10k	1	
R7	10k	1	
R8	10k	1	
R9	10k	1	
R10	10k	1	
R11	10k	1	
R12	10k	1	
R13	10k	1	
R14	10k	1	
R15	10k	1	
R16	10k	1	
R17	10k	1	
R18	10k	1	
R19	1M	1	
R20	10k	1	
R21	10k	1	

Part Number	Value	Sheet	Comment
R22	1M	1	
R23	100k	1	
R24	100k	1	
R25	100k	1	
R26	100k	1	
R38	10k	1	
R39	10k	1	
R40	10k	1	
R41	10k	1	
R42	10k	1	
R43	10k	1	
R44	10k	1	
R45	10k	1	
R46	10k	1	
R47	10k	1	
R48	10k	1	
R49	10k	1	
R50	10k	1	
R51	1M	1	
U1	NEVE BA743	1	P&G fader
U2	NEVE BA743	1	P&G fader
U3	NEVE BA743	1	P&G fader
U4	NEVE BA743	1	P&G fader
C1	20pF	2	
C3	330 μ F	2	
C4	330 μ F	2	
C8	20pF	2	
C9	20pF	2	
C10	20pF	2	
C11	20pF	2	
IC3	NE5532N	2	
IC5	TL074P	2	
IC6	OP275P	2	
IC7	OP275P	2	
QUAD PANNER	Penny & Giles Quad panning joystick	2	
R1	10k	2	
R2	10k	2	
R3	10k	2	
R4	10k	2	
R5	47k	2	
R6	10k	2	
R7	10k	2	

Part Number	Value	Sheet	Comment
R8	10k	2	
R9	1M	2	
R10	100k	2	
R14	10k	2	
R16	10k	2	
R17	10k	2	
R18	10k	2	
R19	10k	2	
R20	10k	2	
R21	10k	2	
R22	10k	2	
R23	10k	2	
R24	10k	2	
R25	10k	2	
R26	10k	2	
R27	47k	2	
R28	10k	2	
R29	10k	2	
R30	10k	2	
R31	10k	2	
R32	10k	2	
R33	10k	2	
R34	47k	2	
R35	10k	2	
R36	10k	2	
R37	47k	2	
R38	10k	2	
R39	10k	2	
R40	10k	2	
R41	10k	2	
R42	10k	2	
R43	10k	2	
R44	10k	2	
R45	10k	2	
R46	10k	2	
S1	on/off/on	2	
S2	on/off/on	2	
U2	NEVE BA743	2	P&G fader
C1	100nF	3	
C2	20pF	3	
C3	100nF	3	
C4	1 μ F	3	tantalum
C5	22 μ F	3	tantalum
C6	20pF	3	

Part Number	Value	Sheet	Comment
C9	20pF	3	
C10	20pF	3	
C11	20pF	3	
C12	1 μ F	3	
C13	220 μ F	3	electrolytic
C14	220 μ F	3	electrolytic
C15	220 μ F	3	electrolytic
C16	220 μ F	3	electrolytic
D1	1N4446	3	
D3	1N4004	3	
D4	1N4004	3	
IC1	NE5532N	3	
IC2	NE5532N	3	
IC3	OP275P	3	
IC4	OP275P	3	
IC5	337T	3	+10V regulator
IC6	NE5532N	3	
IC7	317T	3	-10V regulator
R1	10k	3	
R2	82k	3	
R3	4.7k	3	
R4	100k	3	
R5	22k	3	
R6	10k	3	
R7	10k	3	
R8	10k	3	
R9	10k	3	
R10	10k	3	
R11	100R	3	
R12	100R	3	
R13	100R	3	
R14	33k	3	
R15	680k	3	
R16	100R	3	
R17	100R	3	
R18	5k	3	
R19	5k	3	
R20	4.7k	3	
R21	4.7k	3	
R22	3.3k	3	
R23	3.3k	3	
R24	10k	3	
R25	10k	3	

Part Number	Value	Sheet	Comment
R26	10k	3	
R27	10k	3	
R28	10k	3	
R29	10k	3	
R30	10k	3	
R31	10k	3	
R32	10k	3	
R33	10k	3	
R34	10k	3	
R35	10k	3	
R36	3.3k	3	
R37	10R	3	
R38	1.4k	3	
R39	1.4k	3	
R40	10R	3	
R41	100R	3	
R42	100R	3	
R43	100R	3	
R48	10k	3	
U1	STUDER VCA	3	
U3	Penny & Giles Fader	3	

Please note: components are numbered with respect to each sheet as the Academic Version of Eagle (the circuit design software) does not allow multi-sheet projects. This has therefore been assembled from three separate project folders.

Appendix B

Module Designs

My main instrument is the modular synthesizer. I built my synthesizer using an old flight case and some power supply and distribution parts from Doepfer along with a custom heatsink modification and some legs to allow it to rest at a convenient angle for performance. In vertical or horizontal positions it is uncomfortable to get to many of the parameters.

Figure 5.1 shows the synth in its current format, although modules are rearranged every now and then depending on my approach to a particular piece or performance. This photo is from a rehearsal for *Spiral* and shows the Altec 9067b filters and the Realistic DX-300 shortwave receiver to the right of the picture with three footpedals visible on the floor. The standard Euro format has many flaws, not least of which is the small panel size, forcing controls and input/output sockets into close proximity. The small panel size does however allow for more modules to be fitted into a case, thus allowing greater sound and performance flexibility.

Principal control is via a ribbon controller which is sensitive to both position and pressure. The ribbon controller outputs continuous voltages rather than stepped MIDI data thus allowing a wide range of direct expressive control.

The modules described below have been designed to the Euro format and are either responses to particular needs for my own pieces, or have been necessary to test hypotheses stemming from my research.

- Big Knob Filter
- Studer A80 Varispeed Remote
- Ring Modulator
- Slew Limiter/Bypass

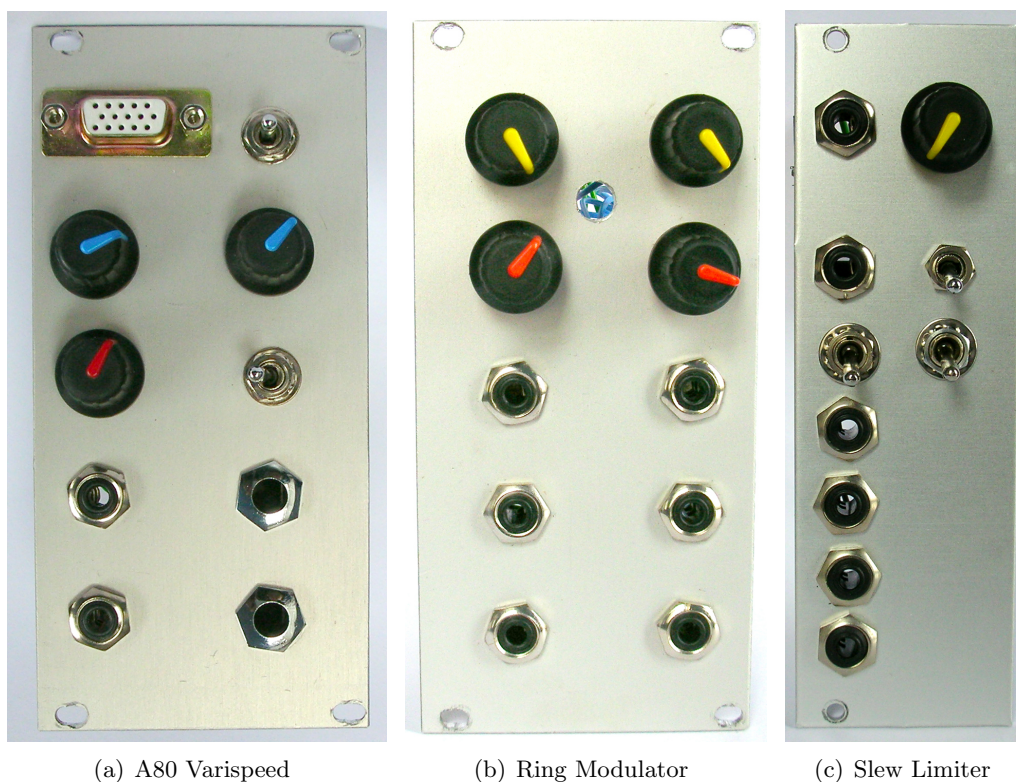


Figure B.1: Selection of self-designed modules

B.1 The Big Knob Filter

This module had to be designed and built in order to research King Tubby's mixing techniques. It started with the acquisition of four 24-position rotary wafer switches, offering the same angle of rotation between positions as the MCI stepped filter. The four-wafer design allowed me to design a four-pole active filter with the capacitors being switched to adjust the filter cut-off frequency.

When I started the project the only information I had to go on was the sound of the recordings. Isolating certain sections containing heavy filtering, I carried out analysis using Logic's EQ plug-in in analysis mode. I looped particular sections, added a narrow band of EQ with high boost and swept through the frequencies until I identified the resonant peak at the filter cutoff frequency partly by ear and partly by using the display. This was a fairly subjective method but I ended up with values very close to those discovered later.

Before my trip to Seattle to examine King Tubby's mixing desk all I knew about the filter design was the frequency data, the number of steps, and the fact that it had quite a steep slope. Having sourced appropriate switches I proceeded to design and build this



Figure B.2: Big Knob Filter, front panel.

prototype quite quickly, relying on numbers of components that I had to hand rather than spending a lot of time and money on particular values of polystyrene capacitor for an extremely high quality signal path. This design was intended to test whether the general sound was anything like the original, and if it was then I would develop a further filter to much higher standards.

Further research revealed a list of frequencies (Veal, 2007, p. 114) and eventually this was confirmed when I received some high quality photographs of the mixing desk from Experience Music Project curator John Seman. The frequency values were displayed around the filter knob (see Figure 3.2). Without having any information about the design of the filter I had to guess based on the sound on the recordings how it might have been designed. I knew that Harned had developed op-amps for MCI earlier than most other companies had used them so I guessed that the filter was an active design using op-amps. The resonance and steep slope of the filter suggested at least a two-pole design, and the phasiness around the cut-off frequency made me choose a four-pole (24 dB/oct) variation with a sharp cut-off frequency and deliberately poor phase coherence - this natural design trade-off worked in my favour. Knowing the frequencies allowed me to calculate component values. The interface was crucial, and it took several months

of searching trying to source some new old stock four-wafer 24-position rotary switches so that I could properly achieve the 11 switch points within 165° of rotation.

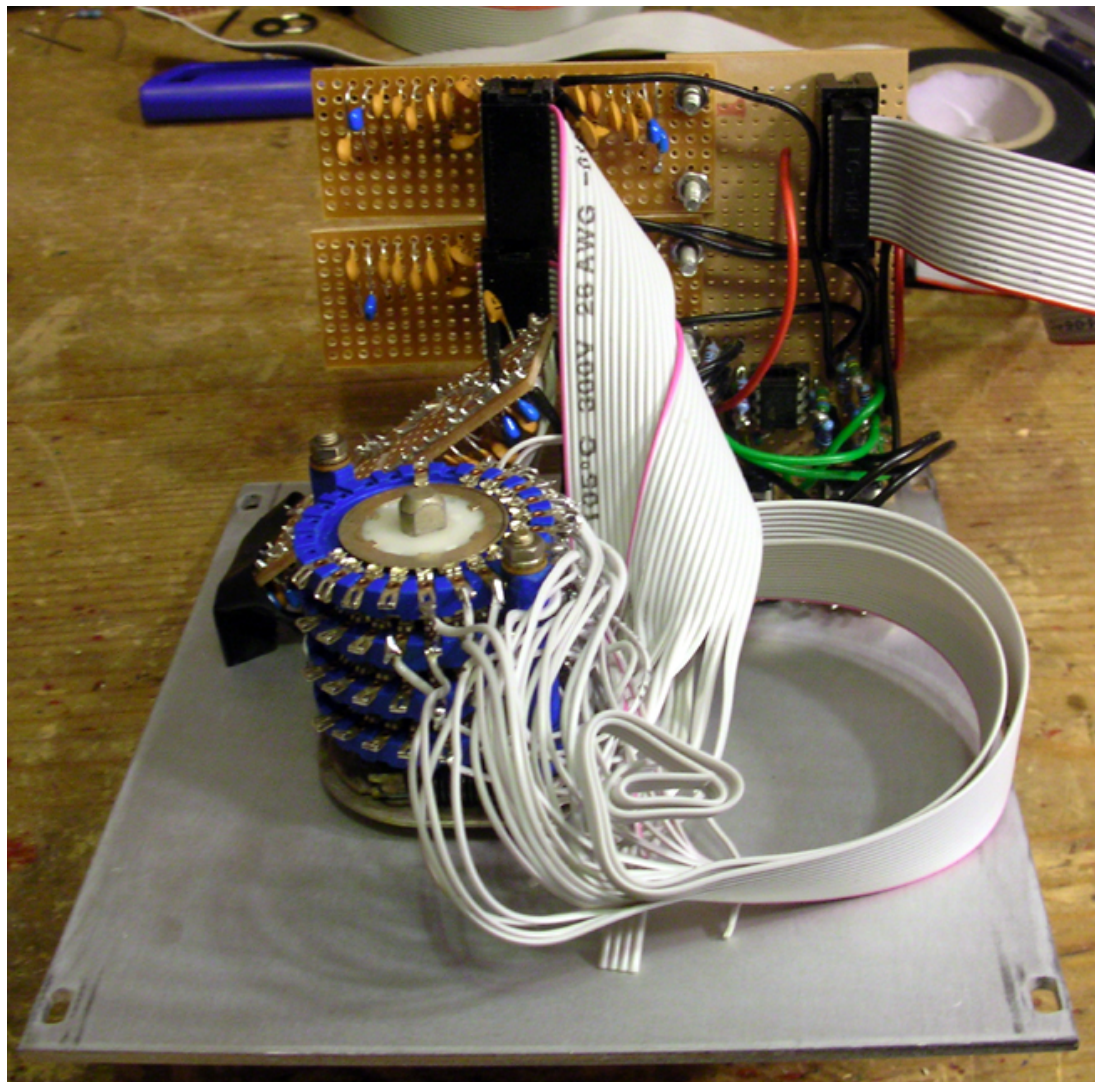


Figure B.3: Big Knob Filter, circuit and switch.

I chose a Sallen-Key design as described by Don Lancaster (Lancaster, 1975, p. 176) since the switched components could all be identical thus saving time in the design and build and troubleshooting. The design calls for some good calibration, especially of the damping resistors so that each pole of the filter is damped to the right amount, thus creating the selected pass band shape. As I wanted a steep slope, I tried to design a 3 dB dip response but the variability and possible instability in my op amp implementation initially caused severe oscillation of the filter. I was forced to replace the carefully calibrated damping resistors with potentiometers, and these can be seen on the faceplate of the module. These must be adjusted by ear so that both filter sections are not quite in oscillation. The benefit of this design flaw is that by introducing these

potentiometers it is now possible to vary the resonance of the filter to suit different needs.

Since identifying Tubby's actual filter and finding an identical model I have been able to work out the component values and circuit used in the original and will be able to construct a replica if needed. Rather than being about precise component values, the subtleties of constructing an accurate replica will no doubt be influenced by the more esoteric properties of the toroidal inductors used in the Altec 9069b such as core material, winding pattern etc. This level of material details verges almost on the alchemical but has a very satisfying connection to the practice of winding transformers with which Tubby and his apprentices were all familiar. It is extremely unlikely but in the spirit of an extreme material research aesthetic, it might be the case that time spent winding coils may yield insights into Tubby's studio practice that would be otherwise impossible to predict.

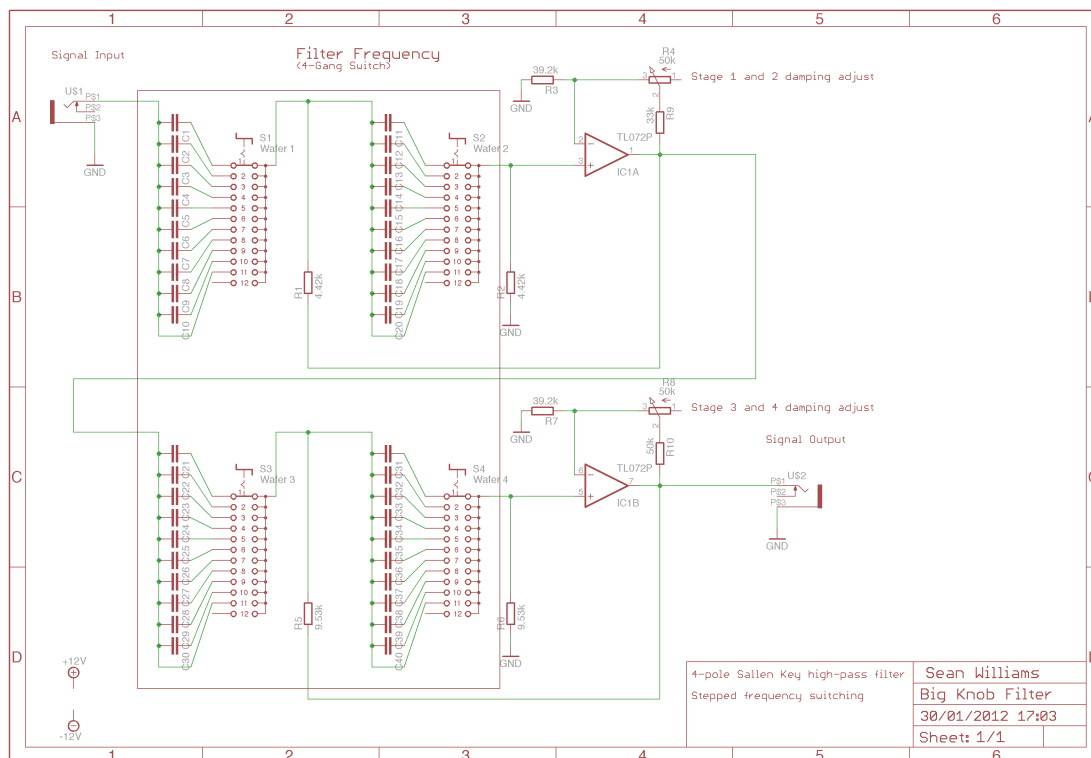


Figure B.4: Big Knob Filter schematic.

Table B.1: Big Knob Filter Parts List

Part Number	Value	Sheet	Comment
C1	8.5 μ F	1	
C2	5.5 μ F	1	
C3	3.15 μ F	1	
C4	2.25 μ F	1	
C5	1.22 μ F	1	
C6	465nF	1	
C7	275nF	1	
C8	180nF	1	
C9	130nF	1	
C10	90nF	1	
C11	8.5 μ F	1	
C12	5.5 μ F	1	
C13	3.15 μ F	1	
C14	2.25 μ F	1	
C15	1.22 μ F	1	
C16	465nF	1	
C17	275nF	1	
C18	180nF	1	
C19	130nF	1	
C20	90nF	1	
C21	8.5 μ F	1	
C22	5.5 μ F	1	
C23	3.15 μ F	1	
C24	2.25 μ F	1	
C25	1.22 μ F	1	
C26	465nF	1	
C27	275nF	1	
C28	180nF	1	
C29	130nF	1	
C30	90nF	1	
C31	8.5 μ F	1	
C32	5.5 μ F	1	
C33	3.15 μ F	1	
C34	2.25 μ F	1	
C35	1.22 μ F	1	
C36	465nF	1	
C37	275nF	1	
C38	180nF	1	
C39	130nF	1	
C40	90nF	1	
IC1	TL072P	1	

Part Number	Value	Sheet	Comment
R1	4.42k	1	
R2	4.42k	1	
R3	39.2k	1	
R4	50k	1	linear pot
R5	9.53k	1	
R6	9.53k	1	
R7	39.2k	1	
R8	50k	1	linear pot
R9	33k	1	
R10	50k	1	
S1	Wafer 1	1	
S2	Wafer 2	1	
S3	Wafer 3	1	
S4	Wafer 4	1	

B.2 A80 Varispeed Remote Control

Although the design of this module is very simple, consisting of a three position switch, two potentiometers and a couple of resistors the biggest challenge was finding a connector compatible with the Studer connector and cable. This is a 14 pin Amphenol connector and it took me some months to track down a source. In the meantime I made a working version by butchering a 25 pin Amphenol connector commonly used on old RS232 interfaces and often found at car boot sales for pennies. By using a standard 15-pin d-sub connector for the other end I was able to use old computer monitor cables to connect the module to the tape machine, meaning I only needed one Amphenol connector for each remote. Eventually I realised that the same connector was used by Roland for their pre-MIDI era DCB interfaces on certain keyboards. Large companies often do not offer good support of older products so instead of contacting Roland I contacted Kenton Electronics who continue to make MIDI/CV converters and MIDI controllers as well as offering MIDI retrofits for older synthesisers. They had some of the right connectors in stock for their own use but very kindly sold me a couple so that I could build two A80 remotes.

The circuit is built around the design shown in the Studer A80 service manual and offers three modes: bypassed, fine adjust of $\pm 3\%$, and coarse adjust of ± 7 semitones. The latter setting is most appropriate when using the remote to control the Studer A80 as a tape delay. This module was built specifically to allow extreme delay time control and modulation for my portfolio composition *Electronic Skank*.

The module is pictured in Figure B.1(a). The lower section features another circuit

that I made which steps down an output signal gain and then provides variable input gain so that guitar effects pedals can be inserted into the signal path of the synthesizer at the correct gain level. This was simply to use up the rest of the panel space but has not been used much and I may adapt the module further to allow it to interface better with the tape machine instead. This feature is not documented here.

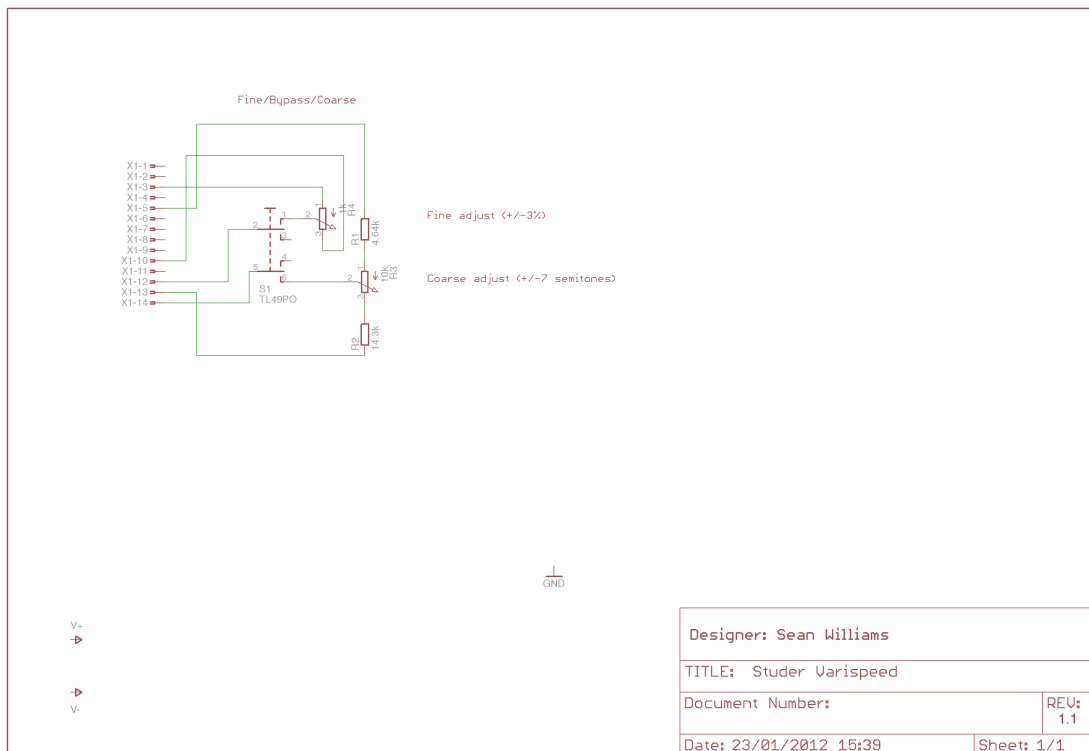


Figure B.5: Varispeed Remote schematic.

Table B.2: Varispeed Parts List

Part Number	Value	Sheet	Comment
R1	4.64k	1	
R2	14.3k	1	
R3	10k	1	linear pot
R4	1k	1	linear pot
S1	on/off/on	1	

B.3 Ring Modulator

This design is based on the very early passive design as outlined by Hugh Davies. It consists of two centre-tapped transformers and four germanium diodes. This design was accomplished in three phases. The first was a basic passive ring modulation circuit with 3.5 mm mini-jack inputs and output. I used a surplus Analogue Systems faceplate for mounting the electronics and to achieve compatibility with the Eurorack format. The second phase was to add three parallel outputs and one switched output. This proved reasonably useful but was superseded by the fourth stage of the design.

Stage three was a duplication of the ring modulator circuit so that two individual ring modulators can be contained in the one module. This stage was undertaken with the idea of creating a frequency modulator or a “Gagaku circuit” (Stockhausen, 1972), both of which demand two ring modulators. In conjunction with this I therefore developed a dome filter also known as a Hilbert shifter which outputs two versions of the input signal with a 90° phase difference between them. In order to carry out some simpler modulation/demodulation (mod/demod) functions I normalised the modulator inputs of each modulator together and normalised the output of one to the input of the other. These connections are broken when another signal is plugged into the relevant input socket.

Since the voltages used in my modular synthesiser setup can peak at up to +/- 10 V I subsequently added variable passive attenuation to the input of each (yellow knobs) and a variable make-up gain amplifier to the output (red knobs) so as not to drive the ring modulator into distortion by overloading the transformers. This relies somewhat on the signal going to the carrier input to have the capability of being attenuated, but since I am using laboratory oscillators as my modulator sources this is easily achieved. Separating the controls at the top from the input/output sockets at the bottom is influenced by the Cwejman designs and allows easier control of parameters without getting entangled with cables.

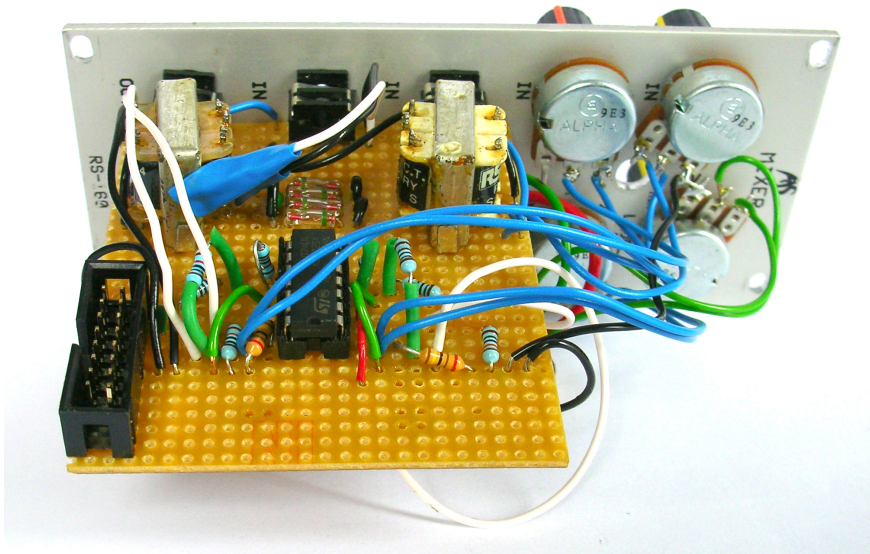


Figure B.6: Ring Modulator components.

The final design of the unit has been used in many performances and features strongly in my portfolio piece *Testing, Testing*. A further development is to build a ring modulator to the same basic circuit design but with much higher quality transformers and germanium diodes matched as accurately as possible. This could then be compared to different generations of ring modulator design including the balanced modulator using an AD633 or MC1494, and a digital version in Max/MSP using the \sim object.

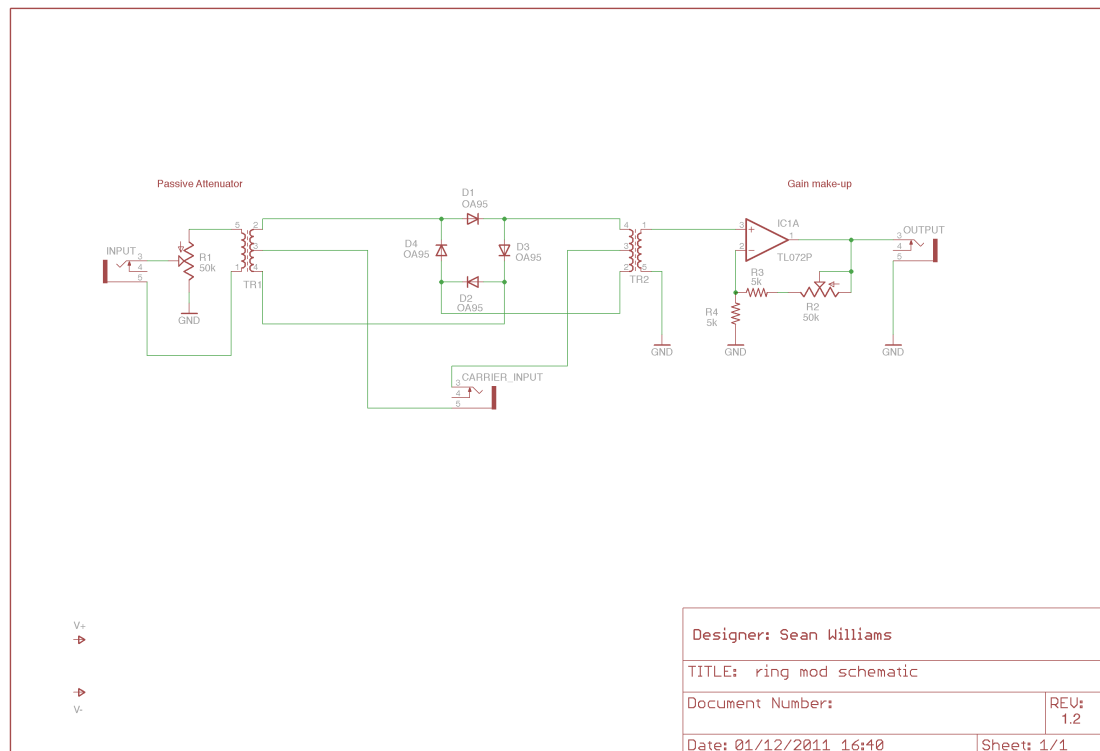


Figure B.7: Ring Modulator schematic.

Table B.3: Ring Modulator Parts List

Part Number	Value	Sheet	Comment
D1	OA95	1	Germanium
D2	OA95	1	Germanium
D3	OA95	1	Germanium
D4	OA95	1	Germanium
IC1	TL072P	1	
R1	50k	1	linear pot
R2	50k	1	linear pot
R3	5k	1	
R4	5k	1	
TR1	1:1 ct	1	10k impedance
TR2	1:1 ct	1	10k impedance

B.4 Slew Limiter/Bypass

A slew limiter or lag processor (a line~ in Max/MSP) is a very useful module traditionally used to create portamento effects. I wanted to experiment with linear and exponential responses so I built a module based on Harry Bissel's linear glide design and a standard exponential glide (Various, 2006). Both these designs are very simple generic integrator designs but adding a switch for linear/exponential was a useful addition.

This module was later customised to include a bypassable insert point, and the additional sockets were also normalised so that the module could handle a CV input, running this through an inserted module (always a voltage quantiser so far) and then the lag processor, either of which may be bypassed by independent switches, thus enabling an extra measure of performance control without the need for re-patching, thus keeping control voltage signal flow intact.

An example of use in *Spiral* would be using the ribbon controller position to control oscillator frequency with quantising bypassed but with some lag. This might enable the creation of radio tuning glissandi, but with a flick of one switch the voltage can be quantised and with another switch the lag can be bypassed, thus enabling tuned semitones to be hit accurately, still using the ribbon controller and without re-patching.

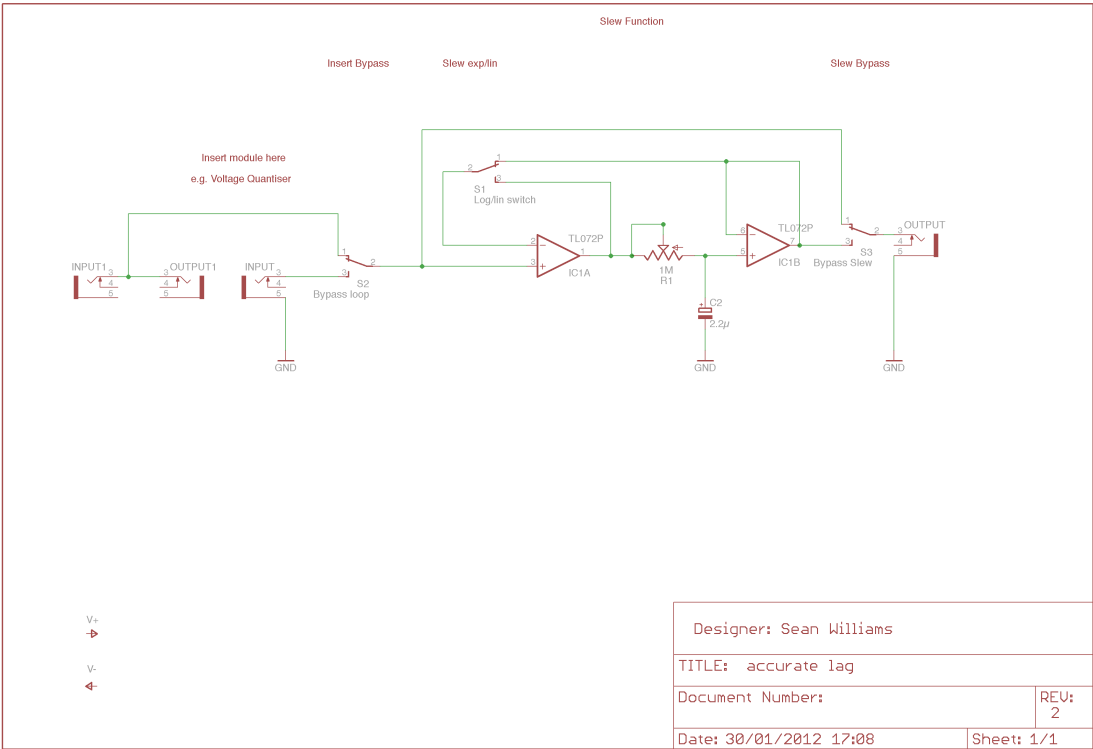


Figure B.8: Slew Limiter schematic.

Table B.4: Slew Limiter Parts List

Part Number	Value	Sheet	Comment
C2	2.2 μ F	1	
IC1	TL072P	1	
R1	1M	1	linear pot
S1	on/on	1	
S2	on/on	1	
S3	on/on	1	

B.5 Other Modules

A number of other modules were developed in conjunction with my practice but since none of them feature heavily in the submitted portfolio work they are peripheral enough to simply be listed.

- Dome Filter - providing two outputs with 90° phase difference between them.
- SSM Filter - based on a Maplin Electronics circuit using a classic filter chip.
- Manual Gate - a very early module that never really worked properly.
- Buffered Switch - useful for distributing clock signals in ensemble performances.

Appendix C

W49 Rebuild

Please see the Portfolio DVD *PhD Materials/W49 Mockup.pdf* for metalwork design sketches and more photographs of the unit.

C.1 W66c Faders

The rebuild of Stockhausen's W49/W66c instrument was entirely based on the fortuitous acquisition of 16 Maihak W66c faders as a job lot. The faders share some characteristics with the W49 filter, although some of the details are a little different. The W49 has a throw of 160 mm where the W66c has only 140 mm, but this is still much greater than the standard 100 mm throw of most contemporary professional faders. The W66c is a balanced design relying on two separate resistive tracks, one either side of the central body of the fader.

C.2 Interface Design

The main requisite was to have two faders in one slot. This was achieved by dismantling two W66c faders and installing a contact runner taken from one on the rails of the other unit. The electrical design was completed by decoupling the flexible contact from one side of the original runner and attaching this to the same side of the new runner. In this way instead of one runner having two contacts, one each side, for a balanced signal, each runner had one contact only, the first runner on the left hand side, the second on the right.

With two runners on one set of rails they could not pass one another and by treating the balanced signal path as a dual mono path I was able to run two separate signals



Figure C.1: W49 instrument mockup, customised W66c on the right.

through the unit and attenuate them with one fader each. The scaling was limited and different for each fader. The minimum position of the top fader could not reach the minimum position on the resistive track, and the maximum position of the bottom fader was prevented from reaching the maximum position on its resistive track due to each fader being in the way of the other. I was unable to recreate the notched steps of the faders which, on the W49, feel like they are clicking into place as they arrive at each new frequency setting.

A tiny detail not exhibited by my rebuild is that the paddle-type fader caps on the W49 are slightly taller than those on the W66c. They are the same as can be found on the Maihak W44 faders, although finding these is almost impossible. For my purposes, the W66c fader caps would suffice since they are still tall enough to be gripped between the fingers.

It was a simple task to design and build a frame to mount the three faders, but this is also different to the original setup in that it is much lighter, and does not have the same slight angle of the W49. The W66c faders are wired with balanced inputs and

outputs so can be used as passive attenuators in the same way as the original faders. My case has no sides as yet and is not exactly road-worthy, but as a prototype testing unit it is adequate.

The essential features of having the controls in an almost identical spacing was achieved reasonably well, and was good enough to allow Rolf Gehlhaar to demonstrate some of his techniques used in the late 1960s for performances of *Mikrophonie I* as described in Chapter 4.4.2 and as demonstrated in the video on the Portfolio DVD *PhD Materials/Rolf Demo.mov*.

C.3 Software Modelling

An effective way of using the unit to control audio was to run sine wave signals through each fader and then use the amplitude, as governed by fader position, to control some other parameter in a software model. The mismatch between fader levels and the scaling problem was overcome using appropriate software rescaling, and a stepped filter was experimented with in Max/MSP. This experiment was not a serious attempt at making a worthwhile model, it was merely a proof of concept. Modelling the noises resulting from changes in filter frequencies would be a non-trivial exercise, and I would place a far higher premium on rebuilding the instrument properly using capacitors, inductors, and the right kind of switches.

Appendix D

Sine and Tape Study Materials

D.1 Decoding Algorithm

This is the main decoding algorithm used for the portfolio piece *Sine Tape Study*.

```
(let* ((freqs '(34 2 5 89 2 8 1 13 1 3 2 144 144 21 1 55 1 89
 3 89 8 1 13 144 55 89 34 55 21 34 13 34 8 34 3 5 13 1 89 2
8 144 144 89 144))
      (lengths '(8 3 6 2 3 1 5 1 4 9 2 1 3 1 2 5 7 8 10 4 2
12 2 2 1 3 4 2 3 1 4 2 3 3 2 11 6 5 1 4 6 9 7 1
1))
      (spaces '(27.5 1 7.5 10.5 .5 44.5 17 17 2.5 27.5 10.5
1.5 .5 7.5 4 1.5 2.5 10.5 1 17 4 17 .5 1.5 .5 2.5
1 .5 .5 2.5 4 1.5 1 1 2.5 72 4 1.5 .5 7.5 1.5 7.5
4 10.5 7.5))
      (baseFreq 5) ;; frequency divided by 100
      (position 0)
      (thisLength 0)
      (prevLength 0)
      (prevTotalTime 0)
      (magic 0))

(with-sound (:output "lisp/sbkw-sines/decode-tape-neumann-l.
aif"
                 :channels 1
```

```

        :srate 96000
        :data-format mus-b24int
        :scaled-to .99
        :statistics t
        :play nil)
(loop for position from 0 to 40 do
  (setf magic (/ baseFreq (nth position freqs)))
  (setf thisLength (/ (+ (nth position lengths)
    (nth position spaces)) 15))
  (sample "audio/Sine_Waves_St_Cecelia/Bounces/
    neumann_take_4_l.aif"
    prevTotalTime
    :srt magic
;    :width 30
    :start prevLength
    :amp 1
    :amp-env '(0 0 .05 1 80 1 100 0)
    :duration (/ thisLength magic)
    :reflect nil
    :distance 1
;    :degree 45
  )
  (print magic)
;  (setf position (+ position 1))
  (setf prevLength (+ prevLength thisLength))
  (setf prevTotalTime (+ prevTotalTime (/
    thisLength magic))))))

```

The first three variables, *freq*, *lengths* and *spaces* are defined manually and list the frequency and length of each sine tone as well as the length of the subsequent space (leader tape). The *base frequency* is manually defined next, in this case, by setting *base frequency* to 5, each portion will be transposed to 500 Hz. The *magic* variable is the key which drives the decoding, setting the sample rate and defining the duration of each transposed section. The amplitude envelope is applied to each iteration of the loop and is set manually by the *amp-env* variable. In this example the recording processed was by necessity a monophonic file *neumann_take_4_l.aif* and the output file was *decode_tape_neumann_l.aif*.

Each monophonic recording would be decoded separately and then the resulting files would be mixed in Logic Pro.

As mentioned in Chapter 6 I have been unable to install LISP/Common Music on my current system, and at the time of writing, it will not run in Lion at all. I am therefore unable to offer any support for the above code.

Appendix E

Rolf Gehlhaar Interview

An interview with Rolf Gehlhaar was conducted at his home on Wednesday 15th June 2011. The interview was in three parts but due to technical problems the first part was not recorded, so this part is documented from notes taken. A short video was taken of Rolf demonstrating some performance techniques used with the W49 filter and W66c fader units on my specially constructed model of this unit. This appears in the Portfolio DVD *PhD Materials/Rolf Demo.mov*.

This appendix contains selected notes and transcriptions from this interview.

Reference was made to the following scores:

- Solo (Stockhausen, 1969*a*)
- Spiral (Stockhausen, 1973)
- Prozession (Stockhausen, 1969*b*)
- Mikrophonie I (113-126) (Stockhausen, 1974)

E.1 Interview Notes

E.1.1 Performing with the Stockhausen Ensemble between 1966 and 1970

Although not studying music at the time, Rolf was eager to attend the seminar by Stockhausen held at the University in 1966. He was familiar with *Kontakte* (which remains one of his favourite pieces of electronic music) and was invited to the rehearsals for this piece by Stockhausen taking place in the following days. At one point in the rehearsals he chipped in with critical advice to the players on timing and received a

look from Stockhausen but not disapproval. After the session Stockhausen mentioned that he usually had an assistant to help him on these sessions and subsequently Rolf was invited to come to Germany and be employed as the new assistant.

He had to get his father to help pay for the airfare but Stockhausen thereafter provided food and lodging and a wage. Rolf took over the duties after Hugh Davies' departure, and the first performance role, after only six rehearsals was operating the filter and potentiometers in performances of *Mikrophonie I*. An early innovation was Rolf's customization of the devices by way of connecting the W66c faders to the W49 filter using two brass strips. Previously the light weight of the faders made them too mobile and it was difficult to control them all together in performance. The credits in recordings of Mikrophonie of Hugh Davies and Jaap Speck as operators of filters and potentiometers for the same group [bears out the suggestion that the units were not connected together until Rolf's innovation as it would have been almost impossible for two players to control them had they been bolted together.]The photographs in the score show them as Rolf designed them.

Another early role in the ensemble was that of microphonist to Alfred Alings' tamtam playing. In a similar way to his comments and interventions in the Kontakte rehearsals in America, Rolf relates how he started using the resonator (a plastic bowl, cardboard tube, glass etc) to add sound material to Alings' events, using it to excite the tamtam directly, not only elongating such events but, after an approving look from Stockhausen, playing events directly, often after a nodded cue from the composer. Alings was more of a trained ensemble/orchestral percussionist and partly due to other commitments on Alings' behalf, Rolf took over as tamtam player and therefore had to develop a technique for holding the microphone at the same time as being able to use resonators and other items to excite and damp the tamtam. The microphone signal would always be routed through a W49 filter and one or two potentiometers for all performances, as would the signals from the piezo (contact) microphone attached to the bridge of Johannes Fritsch's viola (see E.1) . These would usually be controlled by Stockhausen in ensemble performances but by various other people during the 1970 World Expo which will be discussed below.

Harald Boje at this time bought an Electronium which was described as an accordion keyboard with some electronics attached to it instead of bellows. Rolf and the others called it the "Torturium" because of its rather horrible sound. At the same time Boje bought a Fender amplifier [photos show an amp similar to a Fender but inconclusive] and the most effective sounds he made by kicking the amplifier to agitate the spring in the spring reverb tank. He soon found that the most suitable style was to play very low and/or very high frequency material so that he could be heard and occupy his own space within the ensemble [a technique which we have rediscovered within improvising



Figure E.1: Original contact microphone as used by the Stockhausen Ensemble.



Figure E.2: Contact microphone with clip to attach to viola bridge.

ensembles such as the Monosynth Orchestra and Edimprov. Indeed, this technique has not only been useful when combining electronic and acoustic instruments, since the extremes of the frequency spectrum are also useful when playing in exclusively electronic ensembles. The logical extension of the low frequency extremes are easily exploited with analogue synthesis methods resulting in a smooth transition between low frequency pitch and rhythm.]

At first the ensemble played pieces by Stockhausen as well as by others, including Johannes Fritsch's *Partita* for amplified viola, filter and tape delay, *Modulation* for tape, and Michel von Biele's tape piece *Fassung*.

Solo was mentioned with great importance with respect to Gehlhaar's own compositional contributions to Stockhausen's work. This piece Gehlhaar referred to as "cursed", carefully choosing the word. Rehearsals of the piece by were problematic and

the piece did not seem to work. Around the same time, Gehlhaar and Stockhausen travelled by train to Basel to see Heinz Holliger rehearsing the piece and they felt that his version was also not working, with the problems lying more within the piece itself rather than in the interpretation. On the train on the way back Gehlhaar suggested the inclusion of periodicity into the +/- notation. This was a technique that he had been employing in his playing within the ensemble anyway, but it was absorbed into *Prozession*, *Spiral*, and other pieces.

E.1.2 Expo 1970

Stockhausen was not present for much of the time so sound projection was done mainly by Mesias Maiguascha, David Johnson and Rolf Gehlhaar. The notes to *Spiral* and the liner notes from the Stockhausen Verlag CD of *Spiral* and *Pole* indicate that Stockhausen did sound projection for these pieces with occasional assistance from Maiguashca...

Rolf's performances of *Spiral* had to be about 12 or 13 minutes in duration and as he had to play it so many times [how many?], he got into the pattern of playing it the same each time rather than strictly adhering to the score when it demands that each *Spiral* event be used as a start point for each subsequent performance etc. Rolf generally got to the end point in the middle of the second line of score. He stressed that because the performance had to be musical, it was often the case that he would get into something and keep playing in a certain direction without necessarily following each indication, and then he would rejoin the score at a particular event. He showed an event on the first line with a joined up bracketed section of transformations as an example, after which a *Spiral* or expansion or AKK event would be the cue to rejoin the score.

E.2 Selected Transcriptions

E.2.1 W49 Performance Practice

Sean Williams (SW): I think one of the details you showed me was that you don't actually have to either move, the difference between the front and rear fader doesn't have to be too extreme.

[The following short section was filmed and is included as Portfolio DVD *PhD Materials/Rolf Demo.mov*]

Rolf Gehlhaar (RG): Well you see the nice thing is, you would always, I would always, I would say, when I set the thing up I would balance it in such a way, that I would

hold these things like this, right, and I would balance it in such a way that forty [dB] in the back and say, twentyfive [dB] in the front would be even distribution, OK? SO if I wanted to make the sound flow, I could do just this by turning my hand, or I could do this - bring it up louder in front, and not change the back very much. The back - you'd have maybe eighteen or twelve dB difference between front and back, whereas in the front you could all the way down, because they would still be there.

SW: because of the acoustic sound you mean

RG: Yeah. You would do it sometimes to chop the sound. You know, you'd do that, but you could also do that. But the thing is that one [front] was much less effective than this one [rear], because people face forward. It's obvious. It's simple.

SW: And the grading on the faders, these faders, it's not linear so you must have developed a bit of a feel for where it starts to go more exponential.

RG: Yeah, you're just feeling and also it depends on the acoustics of the hall and how much oomph you've got, you know. Some halls we couldn't play it loud because they were too reverberant so you keep it down at chamber music level. And other places you know, you could slam it.

SW: Could you just, earlier on you were demonstrating that to get rid of the clicks sometimes you'd ...

RG: OK, so if you had it like this, you had it like this and then I would go [reduces front fader quickly, changes filter fader, and brings front fader back up fast].

SW: And you'd do that with both the front and back, you pull back both faders?

RG: Always, every time you'd move this [filter faders] you bring it down a bit, move this, bring it back up, but just really quickly, so all you would hear in the sound but just be a little wobble, but it would take away the clicks. Because they were, sometimes the clicks were... I'm trying to remember, sometimes they were the worst at low frequencies. You could hear them when the tamtam was in its lower registers making low sounds, and, and sometimes you wanted to squelch [emphasize] them by taking these two up all the way, so then you would [action] and that's what you would do right. And you'd just do it a little bit like this and to hell with the score. Because in the end what you wanted to do was to get that ideal ... that you had... when you heard it, and you kind of knew what it should sound like, and then Karlheinz would say "Well you know this has got, try to make this sound a little more distant, you know, so take away the highs, you know, or this one I want it to be quite squelchy and close so, you know, make sure that you get up there, you know," because sometimes, it's hard work, you know, it's really hard work.

SW: Because they're heavy levers. This double thing I've got here is just a lightweight

lever.

RG: They're heavier therefore the advance is heavy and you sat here like this, right, and you were looking there [score] and you were looking there [instrument] and looking there [performers] and trying to figure out the relationship between that [score] and what was going on on the stage, and what you were hearing. Yeah I mean after a while sure, after a while you could hear it, but we, I only had three rehearsals to learn it. It's quite a long piece, about thirty-some minutes.

SW: It's a monster

RG: Yeah, it was scary. And sometimes I was just glad that I didn't get an angry look from Karlheinz, because he was sitting right next to me you know.

SW: So you were sitting next to one another, you wouldn't have been [far apart]

RG: No we sat pretty close to one another.

SW: You were mentioning the other equipment, the microphone you said was a Neumann...

RG: Well the microphones were I think K54 Neumann. They were condenser microphones, they were about this long [six inches], very expensive and they were about this shape [board marker pen]. And they had an XLR on the back [earlier he talked about the Tuchel connector] going to the German standard which was called Tuchel, super expensive which in the end have been shown to be completely useless - all that mu-metal they use on them doesn't do anything - it's proved that - because they thought the mu-metal would demagnetize you know and stop... XLR don't have mu-metal and they work fine.

SW: That would go to its own power supply.

RG: Yeah, we then built a box which we... the technician in the studio [WDR] the electronic studio

SW: Would that be Jaap Speck

RG: No, no, Speck had left approximately the same time when I came, or had just left. They had a disagreement. The exact reason, what was involved I don't know, I don't remember. It was not Müller, it was Schültz. Schültz became the new technician and he designed the preamplification system. SO we traveled with preamps for the microphones, and the filters. They all went into this box, if I remember correctly. Yeah, it had about eight Tuchel ... on the front. You've probably seen it.

SW: I've seen pictures of it. I haven't got any really good close-ups of it.

RG: It had about eight inputs on the front.

SW: It looks like it's got the standard Telefunken V72 modular preamps

RG: Yes, exactly and they were in a chassis that he designed

SW: So the mic would go to a preamp and then to the filter and out of the faders and

because of the, the filter's passive so you're going to lose twenty or thirty dB

RG: Yes they went in there and then to the preamps

SW: so they went back into the preamps and from there to the loudspeakers [power amps]

RG: That's right. And so we would travel with that thing, microphone cables and all of those cables, and then, when we went to a gig without technical support, they would take our output from the preamp to their PA system. Which, when we went with technical support, would be completely passive [in the sense that it didn't have another mixing desk and operator]. Because it would go out of the preamp into amplified loudspeakers. We used to take the Altec Lansings and there was just a cable, and we would govern the output volume on the back of the speaker - set it to seventy five percent or whatever - and do the rest locally.

SW: So you calibrate the speakers individually first

RG: I spent HOURS calibrating speakers.

SW: Did you, at that point, were you using the famous Stockhausen test tape - I suppose you wouldn't have been taking tape machines [with you]

RG: No we weren't. We were taking Stockhausen test ...nothing! Sometimes it became ridiculous. Because you'd be up, like in the Fenice [Venice] or a similar place, you'd be up in the gods, I'd be standing there behind the loudspeaker, Karlheinz would be on the stage and then he'd say "A little bit more to the left" and I'd shift it to the left. He'd say "No, no, it's too much," so I'd shift it to the right, he'd say "No, a little bit more to the left" and then it became a matter of a centimetre, of the speaker there or there, and I would just give up.

SW: He had a bit of a [[hearing problem in his right ear]] (unasked)

RG: A loudspeaker with a cone this size, you know? I mean most of the time he was very good, yes,? But you knew he was taking the piss sometimes. If you spend every day with him for three years there are just times when you know he's taking the piss. And we joked about it.

Um yeah, so technically speaking, we had some adventures. It was always very nice when the WDR sent a lorry out, took all the stuff and sent it ahead because ... we would plan our concerts at sort of two day intervals, and giving the lorry enough time to catch up with us. In those cases where we did our own transport, because we didn't take a PA system, but we had the instruments, which meant we had about eight suitcases and then the tamtam in three parts with the stand and so on.

E.2.2 Microphone Technique

SW: I've thought of another question, an important one actually and it's to do with microphone technique, because you started off using, well some of your role was doing microphone while Alfred Alings was playing the tamtam.

RG: Yes, I did the microphone for Prozession.

SW: Right. Certainly with *Mikrophonie* [I] you've got the different, you've got the distance from, along the surface of the tamtam from the point of excitation and then you've got the distance away from the surface of the tamtam. Now, when you get the microphone too far away you're not going to pick up much at all.

RG: No.

SW: So what I'm interested in is close is fairly close,

RG: Close is as close as you can get without the thing hitting the... [microphone]

SW: and far away, I would imagine far away ...

RG: Not much more than that [demonstrates by holding marker pen next to wall].

SW: We're talking about six inches

RG: Six inches

SW: and so medium about three inches

RG: A bit closer, I mean probably even medium four inches, normal [close?] two inches and for some of the things really close. I mean in *Mikrophonie* there's that business where they hit the tamtam with chains. You don't have to be very close to that because it cuts through. But when you're using a glass or a plastic cup, and you're using the cup both as a resonator for the microphone [and as an exciter] yeah, you know what I'm talking about: [demonstrates] tamtam surface; cup; microphone here; opening and closing it; screeching it; opening and closing it. Right? So there, really close. And in some of these [score 118] "schnelles lafen" but where? [fig 8 pattern] close tamtam? I don't know/remember exactly what these lines mean, close or far away? This would mean horizontal to me.

SW: I think the thick ones are close and on the vertical, this one is close to the point and this is far away but on the surface of the tamtam, so that's kind of...

RG: yeah, yeah, going across the surface. But then nobody's making any sound here anyway so you're just getting the reverberation [118-122]. Yeah I don't see the metronome marking here but this is ... There were some bits that were really really hard to do because they were exceedingly structured, like this. That's quite difficult. 1 2 3 4, 1 2 3, [...] so you've got four different movements here and seven different movements here. And this is, I'm sure the metronome marking here isn't super long (super slow).

SW: But I'm really interested in the fact that you say the maximum distance away

with the microphone, you're talking six inches or so.

RG: I would certainly say no more than six inches.

SW: Because I've seen performances and it's you know [arm's length]

RG: No I haven't, we didn't do that.

SW: and having heard a performance like that and thinking well

RG: You don't hear it. No, because I mean, the tamtam is so strong that the loudspeaker sound has a bit of a struggle sometimes, right, and if it doesn't come out of the loudspeakers then the filters are useless. The beauty of the piece is exactly that polyphony between the amplified and the filtered - that's why the filters are important because you get an amplified sound which is different from the unamplified sound, so it's quadraphonic, or polyphonic.

SW: And the practice that was evolved doing this is something that you took into when you were operating the microphone for Alings as well because you were, in *Mikrophonie* [I] you were doing electronics and then you moved to doing the microphone with Alings and then you were doing your own microphone.

RG: It started like this OK: the microphone techniques that I was using when Alings was playing/we were playing *Prozession*. Because *Prozession* was one voice of an ensemble, I pretty much supported whatever he was doing at the moment with fairly close microphone, unless he was using the big one, you know, and then I'd take it away a bit down on, I would do, he was hitting here with the big thing and I would take it to the edge here and I would scan the edge a little bit because I knew that's where the low frequencies would be, but most of the time I was pretty close. And then, obviously like in *Mikrophonie* [I] I started using a resonator, OK, and then at the same time I started using the resonator as a sound maker (hehe), no matter what he [Alings] was doing, you know, and if he stopped playing I'd keep on. That's when Karlheinz looked - his eyes like this (hehe). And he nodded, you know. And that's when it all started.

SW: And so your practice developed, it was a kind of progression through that so then when you were doing solo tamtam you were also using resonators, not just stimulating the thing, not split up as *Mikrophonie* [I] splits up the roles,

RG: I had both roles. I wasn't doing the filtering and so on

SW: so you were still relying on the totality of the sound to be dictated

RG: manipulated, yeah, by somebody who had experience in what it should/could sound like for *Mikrophonie* [I]

SW: and it would usually be Stockhausen doing that or would it be other people?

RG: No it wasn't Stockhausen because he was only in Osaka, of the six months he was only there about two months, in two sessions, so it would [David] Johnson or [Mesias] Manguashca, Eötvös, you know. And the reason I did the sponge [see earlier note] was because I needed two hands. I couldn't do it because sometimes I wanted to play

polyphonic, right, so I wanted to have a low sound and a high sound and I wanted to do the pluses and minuses like that right?

SW: [referring to *Spiral* score] the arpeggiations, the polyphony

RG: Yeah. I could do [sings an arpeggio] but I needed another hand free so that's the reason I put the microphone on my wrist, and otherwise, I put it in such a way so that if I did this [bends wrist ninety degrees downwards] the microphone wouldn't hit the tamtam, so I could get really close but I knew I would never hit it. So I could bend my wrist and that would be the tamtam and the mic would be really close.

SW: and then presumably that, that obviously then gives you another hand to damp the tamtam because damping is a major factor

RG: Major

SW: certainly in *Mikrophonie* and then I would have to assume that it would have been a major factor in all the rest

RG: Yeah, yeah, dampening was, you learnt to do that with your knee, because you stood, when you were playing it, you usually stood like this with one shoulder to it [stands next to bookcase, 70-80 degrees facing left and using right knee to damp] so it was my right shoulder, and then I would be making sound here and have the microphone here and I dampen it like this. And when I was very naughty I would write something on the tamtam (hehe)

SW: I see, there's a photo on one of the scores for *Prozession* I think and it says something like, well there's the F word on it...

RG: No, I never wrote an F word.

SW: I think there's something and it's written up here in chalk and I can't see exactly what it says

RG: It says Vietnam

SW: Does it

RG: Yeah. That's what I wrote on my tamtam. I did it the first time in Vienna. We did a concert in the Vienna Musikverein, you know, the Mozart... I wrote "Vietnam" on the tamtam and the producer came and said I have to erase that. Couldn't have any politics in the concert hall. We always had chalk because you know, you use chalk. So I said OK, I said to him I'd rather it were there but, you know, I don't want to cause any trouble. So I erased it, and then in the course of the concert, I wrote it again and then nobody could do anything about it, right. (hehe) And there is a photograph of it somewhere where it says "Vietnam" on it.

SW: I'll have another look at that but if I can find it maybe I could email you. If it's not that it would be nice to check it.

RG: Where's that book? No, no, the photograph would be on one of the records... this is the setup for *Kurzwellen*... I was sitting down...

[Note: Rolf confirmed that what he had written on the tam-tam on in the photo-



Figure E.3: Photograph from inside cover of *Kurzwellen* score (Stockhausen, 1969c)

graph from the *Kurzwellen* score (Stockhausen, 1969c) were the words: “I HATE MUSIC”]

SW: And the chalk there is for the glasses and things to...

RG: vibrate better

SW: to get a bit more purchase on the surface

RG: That’s right. I always used a bit of sandpaper, and sanded certain areas of the tamtam with very fine paper. Well, it was our tamtam. People don’t like it if you do it on borrowed tamtams. They all tend to be lacquered.

SW: And did you use rosin and stuff as well?

RG: Yeah. And then I would have one area which was chalk; sandpaper and a bit of chalk, and one area which was sandpapered with rosin, and the things would speak differently on them.

Appendix F

Stepped Filter Taxonomy

Brief Taxonomy of stepped filters.

F.1 Filters

The tone control on a radio is the simplest example of an everyday filter. Fully open (set to 10), the filter allows the entire audio signal to pass through with no effect. As the tone control is turned down, the higher frequencies in the signal are attenuated whilst the lower frequencies pass through unaffected. This common tone control is a low-pass filter with a very gentle slope. Another common example of filtering is the experience of listening to a hi-fi from an adjacent room with the door closed. Most of the high frequencies will be absorbed by the door and wall, but the lower frequencies will still be audible. This is also a low-pass filtering effect but with a steeper slope and a lower cut-off frequency, i.e. the frequency above which the amplitude is reduced.

High-pass or band-pass filters are less common, but the telephone is a ubiquitous example of a band-pass filter. Since the intelligibility of the human voice relies on a relatively narrow band of frequencies, it is possible to discard, or to filter out both very low and very high frequencies without sacrificing intelligibility. Typically a telephone will not reproduce frequencies above 3.5 kHz (a low-pass filter) and not below 350 Hz (a high-pass filter). Since there is a low-pass and high-pass filter in series, we can consider this to be a band-pass filter, i.e. all frequencies within the filter's frequency band are allowed to pass through while all frequencies outside the pass-band are attenuated.

I have grouped these stepped filters into three main categories; American, European, and Laboratory. The lists are not by any means exhaustive but serve to illustrate

the range of manufacturers and the scope of variation in design. A full study would almost merit an entire thesis, but I try to describe the salient features in each group below.

F.1.1 American/Film Sound Effects

This is a selection of filters available both as modules for installation and as standalone units. In general these units seem to emerge in their design from the mid 1930s and the motion picture industry. They tend to feature large rotary stepped switches and are based on passive designs using inductors and capacitors (LC filters). The parallel use of stepped and often rotary controls for volume in the early days of electrical sound recording technology perhaps helps to explain the use of rotary switches for filter frequency, but it is also evident that designing linear, fader-style switches is extremely complicated and expensive and such implementation is only found in a very few European devices.

- Altec 9067b - 19" rack-mount unit incorporating the 9069b high-pass filter and the 9068b low-pass filter in series with switches for bypass and selection of one or the other filter.
- Altec 9068b - low-pass filter module.
- Altec 9069b - high-pass filter module (as used in King Tubby's MCI mixing desk).
- Cinema Engineering 4031B - rack-mount high and low pass filter.
- GE F4A18A1 High & low pass filter, Passive, rack mount
- Langevin EQ-255A - high-pass filter module.
- Langevin EQ-255B - low-pass filter module.
- Langevin EQ-259A - rack-mount unit incorporating the 255A and 255B filters.
- RCA MI 11723 - Sound effects filter, Passive, rack mount
- Quad Eight VFX-200A - mixing desk filter module
- Urei 550/550a - much more restricted range of frequencies: 40, 55, 70, 85, 100, 200 Hz high-pass and 5, 7, 8.5, 10, 12.5, 15 kHz low-pass, intended for mixing or mastering rather than sound effects.

F.1.2 European/Radio Sound Effects

Large budgets made it possible for early filters to be designed around linear fader-style switches. The Hörspielverzerrer devices are the only devices to feature this kind of control, with the Maihak W49 the earliest documented design dating from 1950. The other units in this category are mainly German broadcast units often based on the Danner format. The designation code of the later transistor based units uses the first letter for the type of unit - V for amplifier, U for dynamics, W for filtering or equalisation. The first number is the manufacturer code: 2 for Siemens, 3 for Tab, 4 for Neumann, 6 for Telefunken. The next two numbers retain some consistency with the early valve and passive units.

- Calrec FX 1123 - Telephone FX Unit
- Eckmiller HV-53 Hörspielverzerrer - Fixed frequency low cut and high cut/boost. Not really a stepped filter but included as it is one of the rare items known as *Hörspielverzerrer*.
- Eckmiller HV-55 Hörspielverzerrer - Combined high and low-pass filter module
- Maihak W49 Hörspielverzerrer - Combined high and low-pass filter unit
- Neumann W75 - Combined high and low-pass filter module
- Neumann W475 - Combined high and low-pass filter module
- NTP 182-200 - Combined high and low-pass filter module
- Siemens W293 - High-pass filter module
- Siemens W294 - Low-pass filter module
- Tab W393 - High-pass filter module
- Tab W394 - Low-pass filter module
- Tab W395 - Combined high and low-pass filter module

F.1.3 Laboratory Filters

Most often the laboratory filters are either octave or 1/3 Octave filters and are used in conjunction with measurement equipment. Many models were available and several different parameters could be controlled, sometimes via external slaving to a master unit. Many electronic music studios relied on such filters in the early period before the need was perceived for filters to be used for purely musical ends.

- Brüel and Kjær

- Krohn Hite 3103 - band pass filter.
- Wandel und Goltermann Terzfilter - 1/3 octave band pass filter.

F.1.4 Cinema/Sound Effects Filters - continuous

Although the following filters are not stepped but are continuously adjustable they are included because they fit into the class of filters used for sound effects in radio, film and music, and their use can also be aligned with noise reduction because of the notch filters that are included alongside the high and low-pass filters.

- Lafont LP 21 Cinema Filter Set - high-pass (17 Hz to 200 Hz) and low-pass (1.7 kHz to 20 KHz) filters with three continuous notch filters.
- Urei 565T Little Dipper - high-pass and low-pass filters with two continuous notch filters, designed primarily for removing hum and buzz from cinema oriented recordings. Also promoted as sound effects devices by Urei in some promotional literature, for creating phasing effects.

Appendix G

Maihak W49 Datasheet

Braunbuch datasheet version 1, 5th of December 1951. Translated by Sean Williams. A scan of the original datasheet is included on the Portfolio DVD *PhD Materials/W49 Datasheet.pdf*.

G.1 Distorter for Radio Play Purposes

G.2 Intended Purpose

The distorter serves especially for radio plays the production of acoustic effects by electronic means through frequency cutting. The device contains 2 fader sections. The one with 2 operating knobs allows separate cutting of the low and high frequencies. The second fader allows a more or less strong effect of the selected frequency cutting in 8 steps. In normal conditions the Radio play distorter must be patched between an amplifier with source resistance [output impedance?] of 25 Ohm (V 41) and a fader with input impedance of 670 Ohm (W44).

The device is designed to be mounted on a mixing desk.

G.3 Manufacturer

Maihak AG., Hamburg, after a development of the NWDR. Design implementation 1951.

G.4 Technical Details and Function

The radio play distorter is designed after the principle of the bridge circuit. The impedances are [derived?] from fixed capacitors and three multi-tapped inductors, through which the desired combination of the high and low-pass is made through the appropriate position of the faders. The bridge is terminated through transformers which allow the conversion from 200 to 670 Ohm to be accomplished and at the same time, the whole distorter [unit] to be balanced.

With the low-cut operating, the set position selected accordingly, the low frequency is cut with a loss of around 40 db. The same happens for the high-cut [filter]. With the cross-fader next to the frequency cut controls you can mix in the unfiltered signal so that the distortion effect can be reduced. At this point the attenuation slope is not significant, but only the selection of the attenuation level. The position of the cut-off frequencies (as well as the crossfader) are set by ear and by reference to the data values. The approximate cut-off frequencies for the high and low-pass are concurrently 100, 200, 300, 450, 600, 800, 1000, 1500, 3000, 5000 Hz.

The crossfader has 8 positions. The lowest setting (Setting 1) has the strongest distortion degree [effect level]. In the two end-positions of the high and low-cutters (10,000 and 30 Hz) the frequency response is given through both transformers of the distorter and independently of the position of the crossfader.

The faders are made with commutators [layers], with the tabs being connected by brushes. Each brush is so wide that with each selection it reliably makes contact with three layers at the same time. Also there are 3 neighboring commutators each associated with the 3 components that form part of one setting. This group of 3 layers follows three times behind one another in the same sequential arrangement, so that the dots are joined [connected] through a 9 layer wider field, inside of which it is neutral to itself [it doesn't matter] where the brush stands/is. Next to these 9 layers lie another 9 layers that belong in the same way to the neighboring position. So with switching there is a 3 layer wide field, in which the part of the switch element [is connected?] to the last neighboring position and to another part of the new [next setting]. Thereby this will result in step dip changes in the wave impedance and frequency response, but also possible bypassing through the switch of the tap-selection of an inductor, possibly causing modifications of the attenuation curve. In order to leave the brushes in one changing-position, station/lock the control knob in a strong position.

G.5 Electrical Data

G.5.1 Attenuation

The residual loss is down about 40 db when connecting the distorter between an amplifier of the type V41 and a fader W44 with the crossfader in its normal setting at position 1.

G.5.2 Frequency response

See diagram.

G.6 Mechanical Data

Front plate 450 x 120 mm, slightly sloping Mounting depth 180 mm Weight 10.9 kg
Connected by 8-way Tuchel [T2001]

G.7 Operating Instructions

The type of construction of the distorter is designed mainly for it to be mounted in a mixing desk. The distorter will be connected with the insertion of the 8-pole female connector and can be embedded in the mixing desk without mounting bolts.

Each [strength/setting?] will restrict [limit] the frequency response, so much so that the amplitude will feel lower. If the apparent volume of the distorter is to remain at the same level, the Radio play distorter must be followed by a volume fader so that it can be corrected by ear.

After reduction of the spring force of the contact springs a new spring will have to be inserted, if several readjustments have already been carried out. For readjustment or for cleaning the distorter follow this procedure:

- Unfasten the fader housing (4 bolts underneath).
- Disassemble the housing from the rear.
- Clean.
- Only clean the contacts with a toothbrush, not with grease.
- Put it back together.

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